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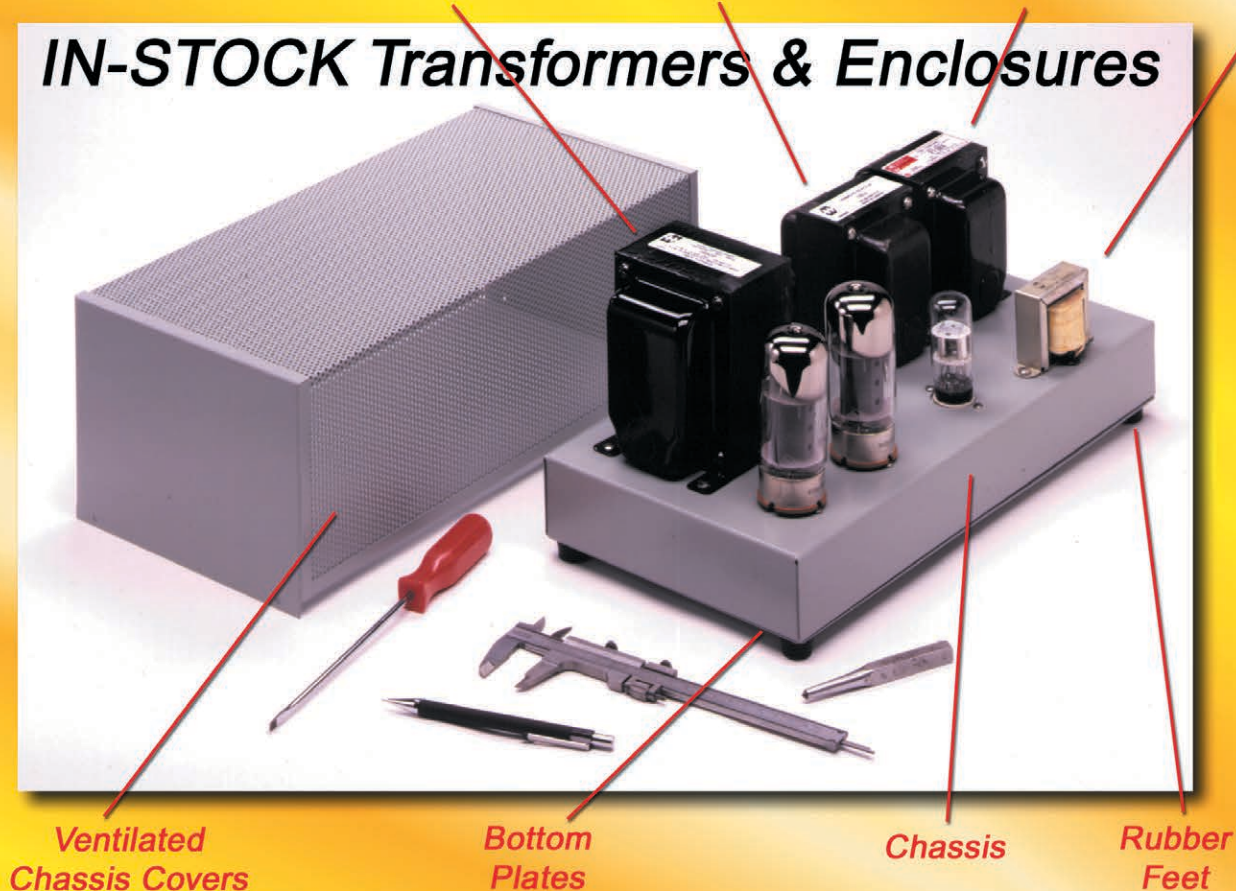
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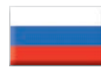
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Audio Showmanship

When you're finished reading this issue of *audioXpress*, you might want to make a note: Tune in next month. In August, we'll be featuring an extensive report on the HIGH END Munich 2013 audio trade show, which draws manufacturers and retailers from around the world.

Ward Maas, owner of Pilgham Audio in The Netherlands, will be your guide through the event, which features many innovative audio products, some of which are not yet for sale. Examples of the broad range of products displayed include:

- The Pivetta Opera Only amplifier designed by Italian Andrea Pivetta. At a maximum 160,000-W output, this is a huge cylindrical amplifier that opens up into sections once the power is turned on. Listeners can literally step inside this high-end amplifier, which has a sci-fi look that suggests it could beam you down to the surface from the starship Enterprise.

- The PureAudioProject open baffle (OB) stereo system. At the polar opposite of the Pivetta in complexity, this item is a DIY kit of matched components for quickly building your own stereo system (including screws, rubber washers, loudspeakers, and high-quality German oak). "PureAudioProject Baffles were designed for simple and easy self assembly, while having OB acoustical guidelines in mind," according to the project website.

Be sure to pick up the August issue to read about these and many other HIGH END exhibitors.

Regards,
Mary Wilson
editor@audioxpress.com

Editor's note: An incorrect biography for audioXpress contributor Ron Tipton was published on p. 44 of the June 2013 issue. To see his correct biography online, please visit <http://audioamateur.com/errata-corrections-updates>.

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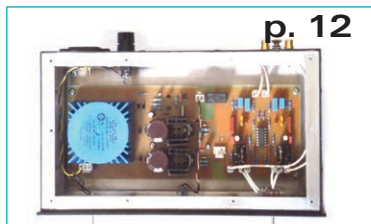
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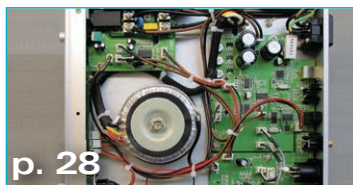


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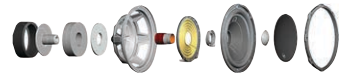
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High-Performance Speakers Meet the HD-Voice Challenge



Technological advances create small speakers with big sound

Most microspeaker and mini-speaker manufacturers started as buzzer manufacturers back in the 1970s. Later the companies expanded into toy and intercom speaker production. At first, performance expectations were minimal and easily achieved. Over the past few years, expectations have changed, beginning with mobile audio products (e.g., docking stations) and continuing with high-quality voice applications.

Mike Klasco's first experience with wideband voice challenges dates back to a 1995 IDEO project and audio experiments for Cisco's first voice over Internet Protocol (VoIP) phone (see **Photo 1**). The VoIP phone was eventually launched in 1997, and it was an instant classic, experiencing great commercial success. The product was conceived as an intranet device with the audio bandwidth specifications under the designers' control. It was not limited by the narrow bandwidth of plain old telephone system (POTS) “landlines.” Without those restrictions, the designers thought it would be easy to provide extended low-end frequencies for natural speech. Wouldn't it be awesome if a speaker-phone could sound as good as a KLH (or Koss or Cue Acoustics) table radio tuned to the news?

VIRTUAL PITCH VS. SPECTRAL PITCH

Through testing, it was determined that clear speech required a 300 to 3,000 Hz bandwidth. This has been the POTS rule for a century. Virtual pitch is the phenomenon in which one's brain interprets musical tones that don't actually exist (in the reproduction of a



Photo 1: Cisco's IP phone utilizes converged voice and data networks while retaining convenience and user friendliness. (Photo courtesy of Cisco)

signal). Specifically, they are phantom “tones” perceived in speech (or some musical production) but missing in reproduction. This contrasts to a spectral pitch, which is a tone that physically exists in the original and in the reproduction.

Paraphrased from *Wikipedia*, virtual pitch is easily visualized by plotting sine waves at several harmonic frequencies and deleting the fundamental, along with any other tones one desires. The resulting wave still has a visible period corresponding to the fundamental frequency (or pitch period) because it is at this point that all higher harmonics constructively interfere.^[1]

It is undeniable that anyone can instantly recognize a familiar voice on the phone—even though the phone is limited to virtual pitch. The term for this is “telephone speech.” Yet, the tinny nasal quality is no match for natural spectral speech (e.g., when Skype correctly works). There is just

too much processing going on in your brain for virtual pitch telephone speech not to be fatiguing.

On the Cisco project, the changes needed to extend the response from 300 down to 100 Hz on the speaker-phone carried implications for sensitivity, back-enclosure volume, amplifier power, and power supply and led to other system design issues (e.g., air leakage, rattles, and echoes back into the microphone). The obstacles seemed insurmountable. Yet over the last year, quite a few projects have targeted these performance goals to meet emerging industry standards.

The driving force behind these projects are the various manifestations of HD voice. There are several emerging standards that extend the bottom-end and top-end frequency responses. For VoIP, the US Telecommunications Industry Association has established standards (e.g., TIA-920A) and similar unified communications standards for software (e.g., Microsoft Lync).

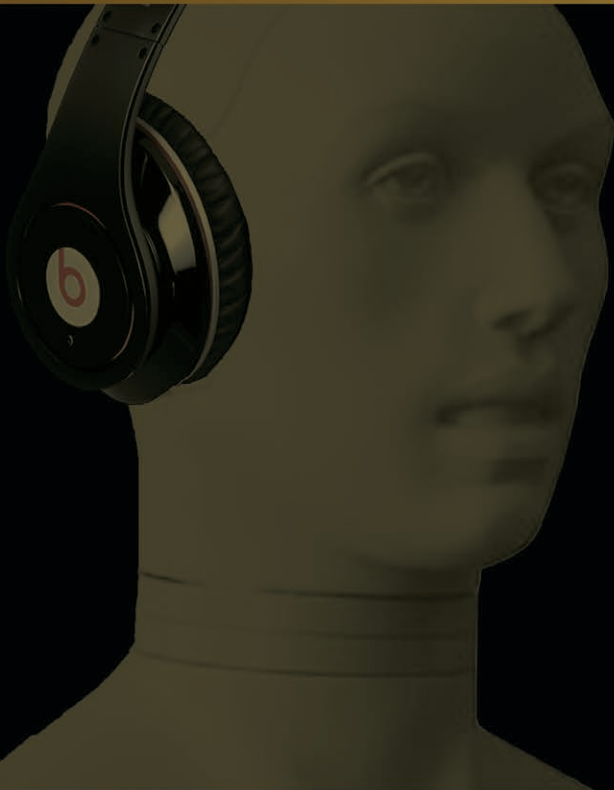
Digital Enhanced Cordless Telecommunications (DECT) telephones have cordless advanced technology CAT-iq2.0 and Dolby Laboratories has Dolby Voice. These systems utilize wider bandwidth and other attributes that provide a cleaner sound quality and reduce listener fatigue.

For communication channels, the upper-end response's wider bandwidth is difficult. But when it comes to speakers, extending the bottom-end creates conflicting design directions. These include back-enclosure volume, speaker depth, diaphragm excursion, product-enclosure leakage, sensitivity, and adequate lower-end acoustic output levels. It is not just

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the speaker driver. The extended bass energy often results in mechanical rattle and buzz in the product's casing and structure-born vibrations, which the microphone picks up. Be cautious when changing wall thickness, adding internal ridges, and compounding plastic filler additives for damping!

MINISPEAKERS VS. MICROSPEAKERS

It's important to understand the difference between a minispeaker and microspeaker. Until recently, the best way to distinguish between the two was by the diameter: a minispeaker's diameter was approximately 2" and greater and a microspeaker's diameter was 2" and less. Today, the lines have blurred and the differences pertain more to the interaction among speaker depth, excursion, and suspension topology.

Minispeakers were scaled down during the docking station craze (e.g., the Altec Lansing InMotion, which featured four 1" inverted-dome full-range speakers with ferrofluid for voice coil centering and heat transfer). Over the last 10 years, the minispeaker's average diameter has shrunk, while the resonance frequency remained low and the excursion remained high.

The typical minispeaker has a two-point suspension—that is a surround and spider (damper) crammed into its tiny frame. The surround is made of a compliant material, while the diaphragm is rigid and made of different material. The voice coil is wound on a bobbin, and there is some space between the voice coil winding and the neck joint where the spider, cone, and bobbin meet (i.e., the triple joint). This space enables higher excursion clearance so the spider does not hit the top plate on backward excursions.

The microspeaker typically has one-point compliance on the diaphragm's periphery. Microspeakers do not have spiders (dampers), resulting in limited low-end and nonlinear excursion capability. The voice coil typically is "monolithic" and self-supporting with no bobbin and limited excursion. Attempts at increasing the

excursion and the compliance cannot control the diaphragm's rocking or the diaphragm from hitting the magnetic structure's top. Several other problems also come into play. The microspeaker's diaphragm is one piece, often a polyester film, but in some applications (e.g., laptops), it is a paper or woven treated fabric. In cell phones, the diaphragm is often made from a laminated composite. For the same diameter, a microspeaker can be half the depth of a minispeaker—so a 1.5" microspeaker might be 8 mm. However, a 1.5" minispeaker with a spider might be 16-mm deep (see **Figure 1** and **Figure 2**).

OTHER MAGIC

The use of ferrofluid is one approach to center the coil, replacing the need for a spider. We discussed this technology in previous articles. Since the spider adds 3 mm or more depth, which often is not available, it's better to provide a "liquid bearing" within the magnet's air gap rather than using a conventional mechanical suspension (spider) above the gap.

The smartphone market has pushed small speakers to their limits. To get the most out of these microspeakers, advanced signal processing enables these speakers to operate at their excursion and thermal limits to achieve maximum acoustic output. Many integrated circuit amplifier companies compete with smartphone signal processing solutions. We think these technologies are also going to migrate to other products including Bluetooth speakers, corded house phones, conference speakerphones, docking stations, and other audio products that must be compact yet deliver room-filling sound (see **Photo 2**).

The ingenious techniques used in today's smartphones are sophisticated and bear a striking similarity to the loudspeaker management signal processors found in concert sound systems. While various HD-voice standards require full performance up to a specified loudness/distance, much of the advanced signal processing

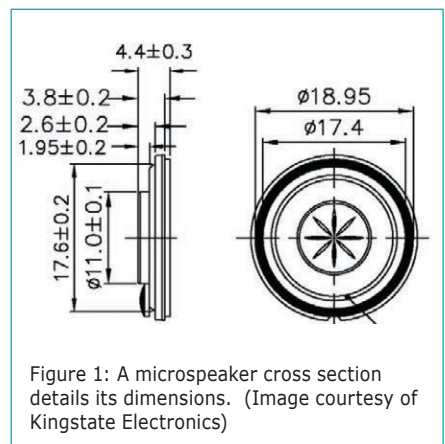


Figure 1: A microspeaker cross section details its dimensions. (Image courtesy of Kingstate Electronics)

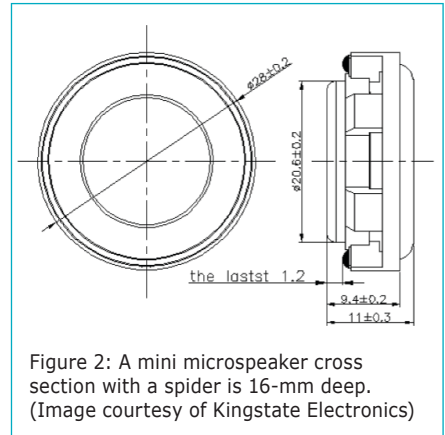


Figure 2: A mini microspeaker cross section with a spider is 16-mm deep. (Image courtesy of Kingstate Electronics)



Photo 2: The ClearOne Chat 50 VoIP speakerphone connects to a variety of devices and provides crystal-clear, hands-free audio communications. (Photo courtesy of ClearOne)

developed for smartphones surpasses sound-level requirements.

Next month, we will take a closer look at these sophisticated processors and the implications for mobile audio, subwoofers, soundbars, and autosound. [aX](#)

REFERENCE

[1] Wikipedia, "Virtual Pitch," <http://en.wikipedia.org/virtual-pitch>.

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Build an External Phonograph Preamplifier

Design modifies modern equipment to play vinyl records

Many of today's receivers and preamplifier processors do not have phonograph inputs, so anyone with a vinyl collection will need an external phono preamplifier. My friend found himself in this predicament and asked me if I could build him one (see **Photo 1**). As an aside, my Outlaw Audio Model 950 preamplifier processor doesn't have a phono input either.

CHOOSING A DESIGN

Texas Instruments (TI) has a series of high-performance op-amps specifically designed for high-fidelity applications. The datasheets include the schematic for a phono preamplifier, the LME49740, which looked like a good candidate. The LME49740 is a 14-pin integrated circuit (IC) containing four op-amps, which is just the right number for the preamplifier. I've had excellent experience with this series and decided to use it.

The first thing I did was design a power supply worthy of the job. I started with a toroidal transformer to minimize the stray magnetic field. Then I used a bridge rectifier with large capacitors to form the raw supply. I regulated the raw supply down to ± 14 V. I used adjustable regulators (the LM317 and the LM337) because they are said to have less noise than fixed regulators. The formula for the regulated voltages is as follows:

$$V_{OUT} = 1.25 \times \left(1 + \frac{R_{210}}{R_{208}} \right) + 0.00005 \times R_{208}$$

Values of 470 Ω or R210 and 470 Ω for R208 provide a 13.985-V output.

The raw supply provides power to a relay driver circuit with about a 0.5 s turn on delay and almost instantaneous off delay. The relay shorts



Photo 1: The passively equalized phono preamplifier contains moving coil capability.

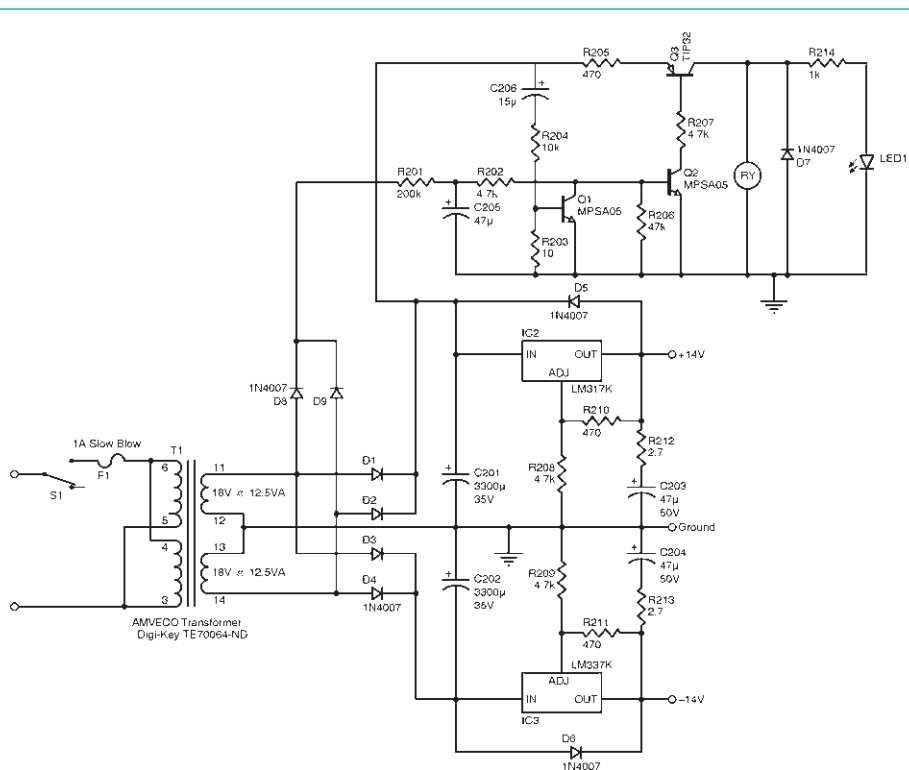


Figure 1: This schematic details the power supply and the relay driver circuits.

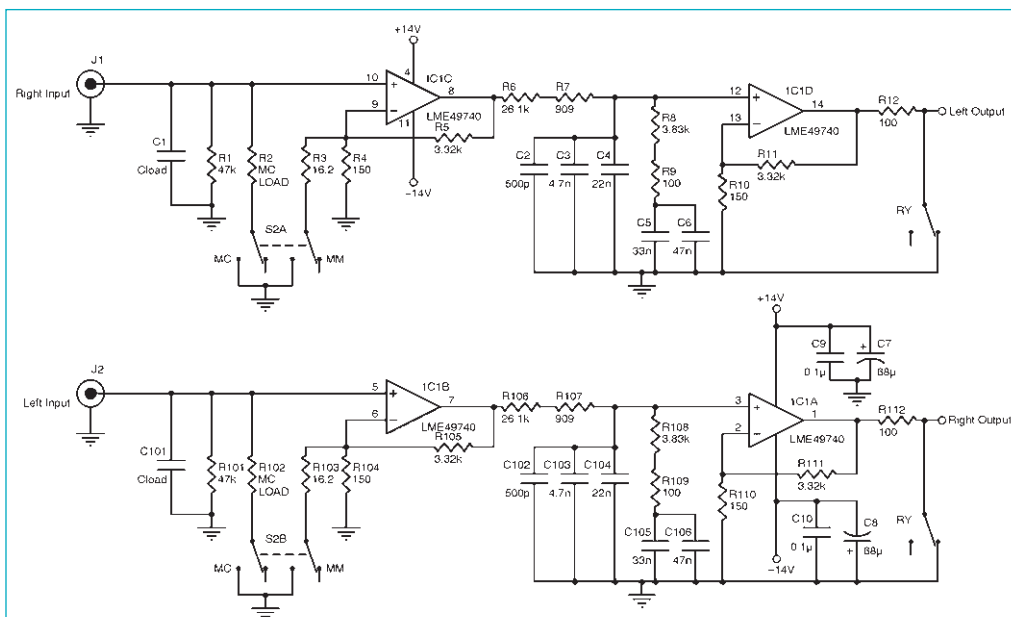


Figure 2: The preamplifier schematic shows changes made to accommodate the moving coil cartridges, cartridge loading, and output muting.

the op-amps' outputs while the power supply stabilizes after power up and at power down before the power supply bleeds off. This mutes the outputs to eliminate turn on and turn off pops. **Figure 1** shows the power supply and relay driver circuits. (The relay plugs into a standard 16-pin dip socket.)

PREAMPLIFIER CIRCUIT

The circuit for the preamplifier itself is essentially the TI-supplied circuit. I've made a few minor changes to accommodate the moving coil cartridges, cartridge loading, and output muting.

Figure 2 shows the modified circuit. The topology is simple. There is a linear gain stage at the input, followed by a passive Recording Industry Association of America (RIAA) filter, followed by a linear output gain stage. The passive filter consists of resistors in series and capacitors in parallel to closely achieve the proper component values. I used the values specified in the datasheet with excellent results. (The TI LME49740 datasheet is available at www.ti.com/lit/ds/symlink/lme49740.pdf.) It includes the phono preamplifier schematic with specified values. Other combinations are valid. A comprehensive explanation of passive RIAA filters can be found on Wayne Stegall's website

(<http://waynestegall.com/audio/riaa.htm>).

CARTRIDGES

Moving coil (MC) cartridges typically have only one-tenth the output of moving magnet (MM) cartridges. Also, the recommended resistive loading is significantly lower. MM cartridges almost exclusively have a 47-kΩ recommended load while MC cartridges have a variety of recommended loads that go as low as 3 Ω. Switch S2 will not only select the proper loading but it will change the gain. The input stage's gain in the MM position can be calculated by the simple equation:

$$\text{Gain} = \frac{(R4 + R5)}{R4}$$

This calculates to 23.13. In the MC position of switch S2, a 15.8-Ω resistor, R3, is placed in parallel with the 150-Ω resistor, R4. The parallel resistance is 14.29. This causes the gain for the input stage to increase by approximately a factor of 10, or to 233.13. A gain that high would be problematic for general-purpose op-amps. But with the LME49740 noise is not measurable with my equipment. The same outcome can be achieved with 10:1 step-up transformers, but they are expensive and I don't think they would be any better.

The recommended loading capacitors for MM cartridges vary greatly. Data for an extensive list of cartridges (MM and MC) is available at www.vinylengine.com/cartridge_database.php.

Figure 3 shows a printed circuit board (PCB) layout that includes the power supply and the preamplifier. I provided component carriers for the cartridge loading components. These plug into dip sockets, so the components can be changed without soldering to the PC board. Since the 47-kΩ load is pretty standard for MM cartridges, I soldered it directly to the PCB. I repeated the same procedure for the gain resistor for the MC cartridges. (If a "high-output" MC cartridge is used, the gain-changing resistor would need to change. However, I didn't include a plug-in provision for that.)

The capacitors specified in the parts list are available from Digi-Key with

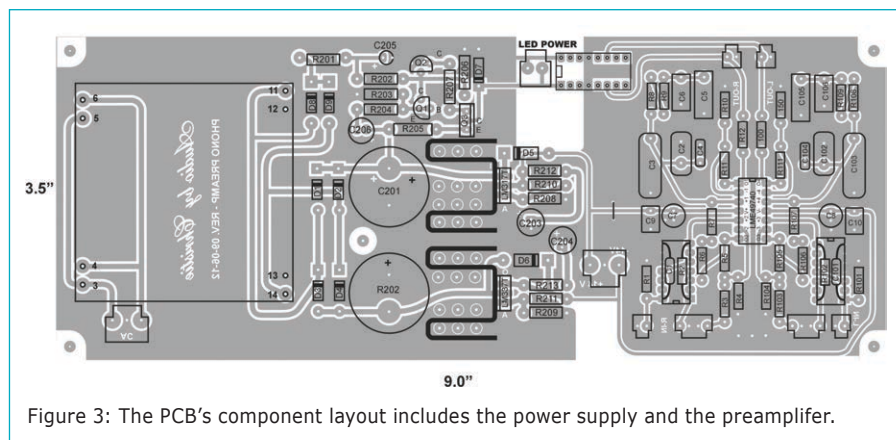
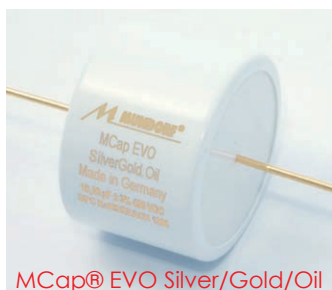


Figure 3: The PCB's component layout includes the power supply and the preamplifier.



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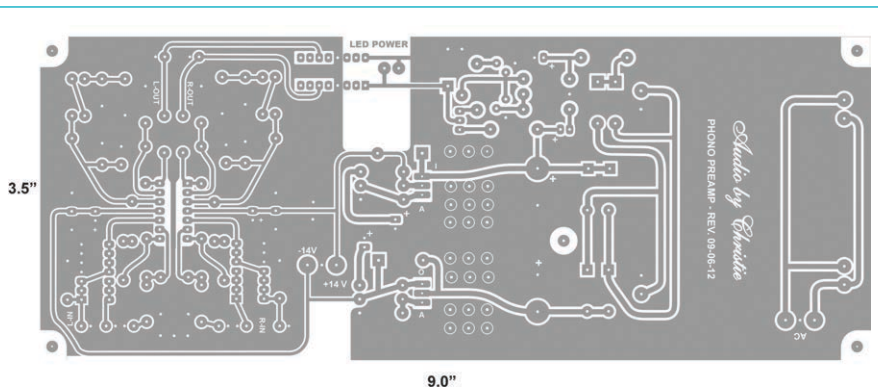


Figure 4: The PCB foil side shows the Molex connectors, which are optional.

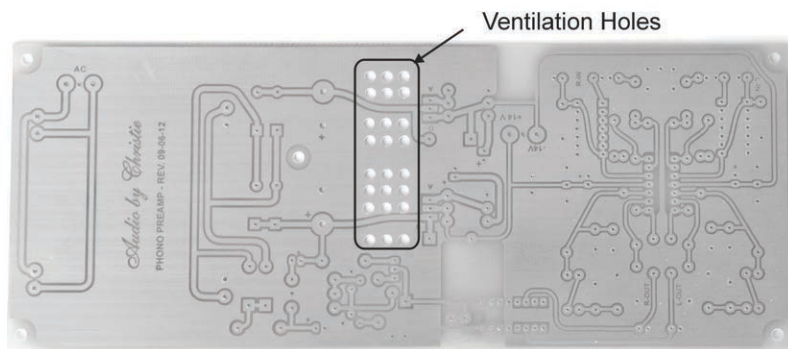


Photo 2: I drilled 24 holes in the PCB to ensure the components are well ventilated.

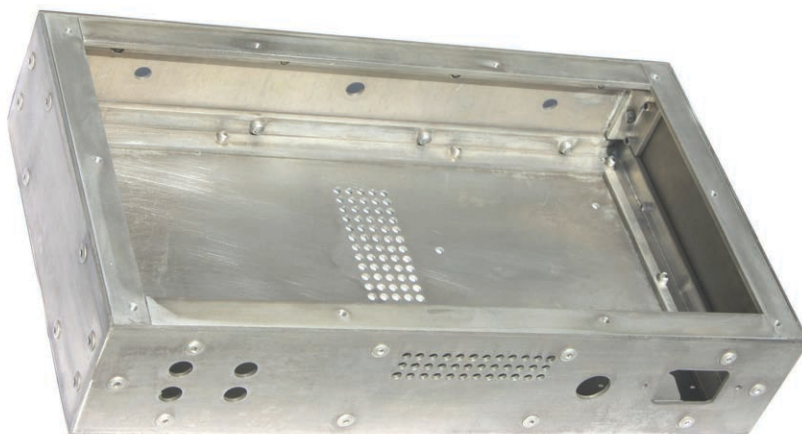


Photo 3: The chassis was a blank slate prior to painting.

lead spacings that fit the board layout. Molex connectors are optional. If used, they will eliminate the need to solder external wiring to the board. I used 0.156" headers for power connections and 0.1" headers for the signal. I provided a header to power auxiliary equipment, if desired. (I am thinking of incorporating this design into a complete preamplifier with input selection, tone, balance, and volume controls.)

Although the preamplifier's power

consumption is quite low and heat should not be a problem, I decided to provide significant ventilation. I drilled 24 holes in the PCB under the heat-sinks for the voltage regulators and array of holes in the bottom and back of the chassis (see **Photo 2**).

THE CHASSIS

I made the chassis out of 0.0625" thick aluminum plates fastened together with 0.5" aluminum angle and pop rivets. The front panel was made

Frequency	Measured	Ideal	Error
20 Hz	19.24 dB	19.25 dB	-0.01 dB
40 Hz	17.79 dB	17.79 dB	0 dB
100 Hz	13.1 dB	13.08 dB	0.02 dB
200 Hz	8.29 dB	8.22 dB	0.07 dB
400 Hz	3.86 dB	3.78 dB	0.08 dB
1,000 Hz	0 dB	0 dB	0 dB
2,000 Hz	-2.55 dB	-2.5 dB	-0.05 dB
4,000 Hz	-6.54 dB	-6.52 dB	-0.02 dB
10,000 Hz	-13.96 dB	-13.73 dB	-0.23 dB
20,000 Hz	-20.15 dB	-19.62 dB	-0.53 dB

Table 1: I tested the preamplifier's response at several frequencies and compared them to published response results.

from a 0.125" thick aluminum plate held to the chassis by the switches. **Photo 3** shows the bare chassis before painting.

Labeling the front panel was

frustrating. I tried silk screening but after three attempts I was not satisfied with the results. I finally used a company that specializes in custom graphic items (Enlarging Arts in Akron,

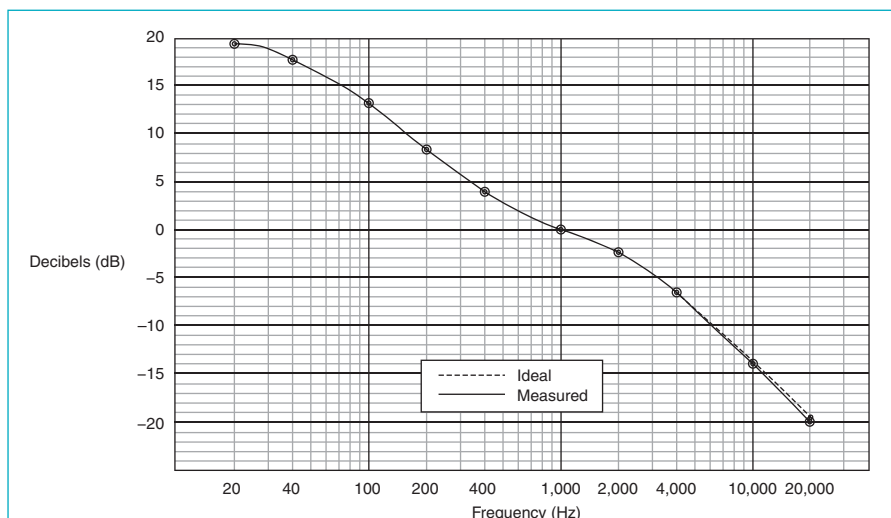


Figure 5: Here is a graphical representation of my test results.

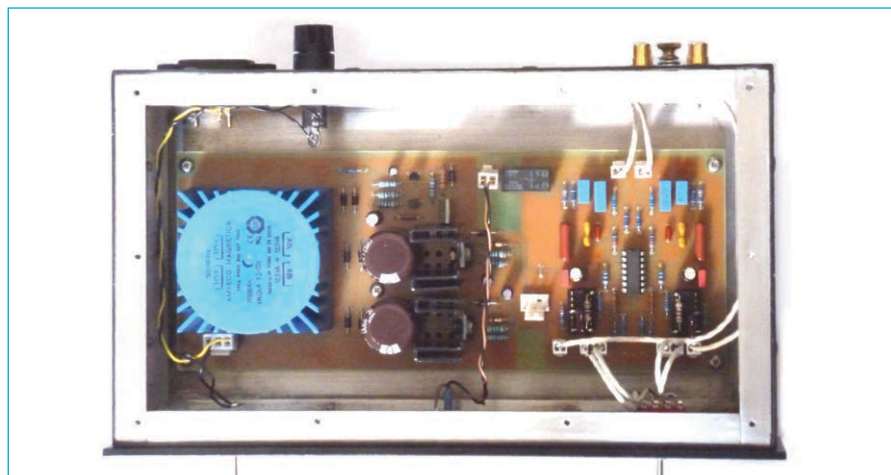


Photo 4: This is an inside view of the completed preamplifier.

7052PH

PHANTOM Powered Measurement Mic System

NOW 4Hz to 25+kHz

<16dBA >135dBSPL

IEC61094-4

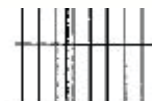
Compliant

1dB/div

30 kHz



Anechoic Free Field Response Included



Rugged STAINLESS Body

Operational

<-20 to >70C

Storage

<-30 to >85C

PHANTOM

<18Vdc

to

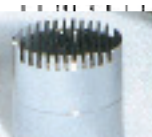
>56Vdc

Removable

Capsule

Titanium

Diaphragm



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Phono Preamplifier Parts List

Capacitors

• C1, C101 = MM Load	Value dependent on cartridge
• C2, C102 = 500 pF, 5%	Digi-Key Part # 338-1069
• C3, C103 = 4.7 nF, 5%	Digi-Key Part # P15431
• C4, C104 = 22 nF, 5%	Digi-Key Part # P4517
• C5, C105 = 33 nF, 5%	Digi-Key Part # 495-1307
• C6, C106 = 47 nF, 5%	Digi-Key Part # 495-1290
• C7, C8 = 68 μ F, 35 V	Digi-Key Part # P12926
• C9, C10 = 0.1 μ F	Digi-Key Part # 445-4756
• C201, C202 = 3,300 μ F, 35 V	Digi-Key Part # 565-2864
• C203, C204 = 47 μ F, 35 V	Digi-Key Part # P5164
• C205 = 4.7 μ F, 35 V	Digi-Key Part # P817
• C206 = 15 μ F, 35 V	Digi-Key Part # P14495

Electrical Mechanical

• RY = DPDT 24-V relay	Digi-Key Part # Z769
• S1 = SPDP Toggle switch	Digi-Key Part # EG2350
• S2 = 4PDP Toggle switch	Digi-Key Part # EG2428
• T1 = Power transformer	Digi-Key Part # TE70064

Miscellaneous

- Molex connectors
- Insulators
- Power cord
- Screws
- Standoffs

Resistors

• R1, R101, R206 = 47 k Ω	Value dependent on cartridge
• R2, R102 = MC load	
• R3, R103 = 18 Ω	
• R4, R10, R104, R110 = 150 Ω	All resistors are 0.25 W 1% metal film unless otherwise specified
• R5, R11, R105, R111 = 3.32 k Ω	
• R6, R106 = 26.1 k Ω	
• R7, R107 = 909 Ω	
• R8, R108 = 3.83 k Ω	
• R9, R12, R109, R102 = 100 Ω	
• R201 = 200 k Ω	
• R202, R207 = 4.7 k Ω	
• R203, R204 = 10 k Ω	
• R205 = 470 Ω , 0.05 W	
• R210, R211 = 470 Ω	
• R212, R213 = 2.7 Ω	
• R214 = 1 k Ω (higher if LED is too bright)	

Semiconductors

• D1-D7 = 1N4007	Silicon diode
• IC1 = LME49740	Digi-Key Part # LME49740
• IC2 = LM317K	Positive adjustable regulator
• IC3 = LM337K	Negative adjustable regulator
• Q1, Q2 = MPSA05	NPN transistor
• Q3 = TIP32	PNP transistor

Ohio). The company made an adhesive-backed decal for the chassis. A durable vinyl film laminated to the front protects the lettering and looks as good as any commercial product I have seen. The cost was reasonable and I would recommend this company to anyone who has a custom-labeling requirement.

In order to test the completed unit, I measured the response at several frequencies and compared the results to published tables. The results for a limited amount of data points are listed in **Table 1**.

The sampling shows a remarkable fit preamplifier, except for a slight drop above 10,000 Hz (see **Figure 5**). The left and right channels tracked almost perfectly. As mentioned earlier, I could not measure any noise.

TESTING THE DESIGN

Having completed the bench testing, a few audiophile friends and I auditioned the preamplifier using a few different phono cartridges. Two of them were MM cartridges, the Audio Technica AT-150MLX with a 100-pF capacitive load and the Shure V15 III with a 400-pF capacitive load. (I chose the low end of the recommended values to allow for the capacitance of the input cables.) I also tested a Benz Micro Gold MC cartridge with a 510- Ω resistive load.

The preamplifier sounded great with all three cartridges, actually better than a highly rated commercial preamplifier. I made two preamplifiers, one for me and one for my friend (see **Photo 4**). *aX*

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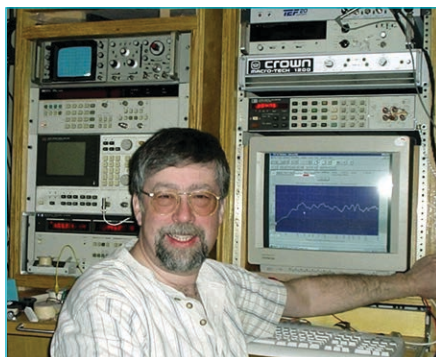
Submit your photos and descriptions to editorial@audioxpress.com. Please include "DREAM SYSTEM" in the subject line.

 @AudioAmateur
  AudioAmateur



Innovator Behind Synergy Horns

Curiosity and ingenuity prove to be a powerful combination



Tom Danley is the Director of Research and Development for Danley Sound Labs.

SHANNON BECKER: Tell us a little about your background and where you live.

TOM DANLEY: I grew up in northern Illinois in an area that wasn't heavily developed, in the same neighborhood where my mom grew up. My brothers and I spent a lot of time outside, playing in the woods and exploring the marsh. We were poor compared to most people in our town, but I wasn't aware of that until school, when I seemed to be reminded of that regularly. To this day, I have a fondness for wildlife and nature, and I prefer the woods as scenery.

My dad was a DIYer and very mechanical. My parents built a structure that had been used as a chicken coop into the house where I grew up. Dad was very handy, skilled at building things. He had a workshop where I spent time watching him, and eventually I was able to use Dad's tools.

By about age 12, I knew how to arc weld with Dad's homemade V-8 powered welder and I had used his metal lathe. At one point somewhere around that same time, I made the transition from taking apart everything I could find to fixing my first radio.

It's funny. Back then, the garbage man, several neighbors, and some of our relatives brought me broken radios and electronic stuff to take apart. My mom

didn't like my bedroom full of electronic parts and the occasional broken vacuum tubes, and so forth, but she allowed it. Dad was a mechanical guy, but he didn't like electricity. I think that may have been my secret ally. Mom had an amazing knowledge of the local nature, history, plants, and animals. And a few times, I thought she might have ESP, which as a teen was kind of creepy. In high school, I got my first job in electronics. It was a part-time job at a local TV store called 20th Century TV. I tested tubes on incoming TVs at first, then later I repaired black-and-white TVs and I put new eight-track players and speakers in cars for customers and a couple of car dealers.

While it was my first real job in electronics, I still think of Mr. Reynolds, my boss, as one of the most fair, even-tempered bosses I ever had. At this time, I was in a band playing bass, building loudspeakers and 200-W tube power amplifiers with repair output transformers from Sunn and other parts from Fair Radio.

Several publications of note turned out to be very important in my life. A quick perusal of my bookshelf shows *Audio Amateur* magazines going back to April 1976, a thick row of *Glass Audio* and *Speaker Builder* magazines.

One reason I agreed to be interviewed is that these publications changed my life in a significant way. No doubt there are countless others whose paths were affected by Ed Dell, and I believe strongly that one thing our country needs are more people who do things, experiment, and invent. Trust me, even a guy (myself) who took five years to graduate from high school can teach himself a great deal. An education doesn't teach curiosity, problem solving, or creativity.

While I have built loudspeakers most of my life and I have occasionally mixed sound, I have also had to

have something significant others have helpfully called a "real job" and that was usually in electronics repair. I had built and repaired a lot of electronics by the mid 1970s, and I expected the job at Grommes Precision, an amplifier company, to be an easy "real job." Funny how wrong one can be. The point being, when you repair something that is broken, you know it is wired correctly but you have one or more bad parts. When you troubleshoot hand-wired tube amplifiers (every one that is produced has to be tested), you find the parts are usually good. If it doesn't work, anything could be wired wrong—emphasis on anything. Anyway, it was a good experience, a completely different kind of troubleshooting than what I was used to, and I eventually got to work part of the time in engineering, but it was too far away.

In 1979, I worked for a company called DSI that made computer tape readers and punches. These long paper tapes that had the instructions on them were used to program CNC machines back then. At DSI, I learned digital electronic logic repair. Compared to analog, that is simple in operation but large on complexity. Anyway, when I was asked to take a crack at the Kleinschmidt Teleprinters and managed to fix one, that became part of my job. These were very complicated machines and while this made them slightly fun to work on, I grew to dread them too.

A co-worker had applied at a small NASA contractor that oddly was right down the street. He didn't get the job so he told me about it. I applied and got the job at Interasonics, a small company that was trying to do acoustic levitation. That is a non-magical way of suspending an object in midair using intense sound. The idea was, in zero gravity, one could position a sample within a furnace and melt/process

it at very high temperatures without any adverse effects from the container. For me, those 17 years were spent in a wonderful environment. One of my bosses was an English acoustician from Mullard Labs who developed the BQS6 sonar transducers. Another boss was a physicist who taught and then worked at Fermi labs.

It was a great opportunity because they really did have sort of a blank sheet of paper and a big list of questions. I was able to find solutions to a number of the problems and I invented a new sound source at 100 times greater efficiency. I became more confident in my work and when I got my first and second patents, I really started to feel like I had overcome my experiences at Deerfield High School.

Eventually, the company grew, I had more inventions, and we flew payloads on several sounding rockets, zero-gravity aircraft flights, and the space shuttle flights STS-7 and STS-51A. My title eventually became Director of Electroacoustic Research. But, the shuttle disaster was the beginning of the end for the company. The work on the space station manufacturing bay stopped and all of that work gradually dried up. I had designed and built large portions of several flight payloads and had the chance to begin astronaut training as a payload support specialist about a year before the accident. I elected not to continue because of my interest in loudspeakers. I wasn't sure about that decision then, but later I was glad I had made it.

SHANNON: How did you become interested in audio electronics?

TOM: I would say several steps led up to it. Around age 5, I heard my grandfather's Heathkit mono hi-fi playing a recording that had French horns. Grandpa lived next door and played a French horn (which didn't make as big of an impression on me), but I was baffled about how the wooden boxes and his stuff with the little light bulbs (tubes) made the sounds. Also, what eventually became just as interesting, I wanted to understand how the music got into the black discs on the thing I was told not to touch (the turntable).

I did chores for Grandpa and in exchange he bought a Heathkit crystal radio kit for me. I'll never forget lying in bed with an antenna wire strung around the room and over my bunk bed, listening to WGN while I was buried under my covers. For most of my life, I had a shortwave radio next to my bed as well, which over the decades resulted in a healthy skepticism of our domestic news media. But it was entirely entertaining and awe inspiring for my two young daughters, who shared my fascination with how the radio worked. (I guess they took after their father.)

At about age 7 or 8, my uncle demonstrated an old telephone's hand-crank generator by giving me a decent shock. That was baffling, how something that felt so strong could be invisible. The shock was a mystery but happened every time. He later gave me the generator. I eventually brought it to school and had my third-grade class hold hands (including me and the teacher) in a circle. It gave us all a mild shock. Electricity was cool! If you hid the wires carefully, you could even electrify swing set chains turning the swing into something like an ejection seat. (I know from experience, as did the neighborhood kids.)

Another turning point in my life occurred around 1974. After high school, I made live sound speakers for TC Furlong and did the sound mixing for bands. I had a bunch of friends who were going to meet at the club where I was mixing, so I thought it would be fun to record the night on my reel-to-reel tape deck.

I plugged into the full-range output from the board, Y'd it into my tape deck, and flipped the tape at break. When I got home at 3:00 A.M. that night, I was too awake to go right to bed, so I set up the tape and listened to it. I was dumbstruck. At home, it was near record quality. It sounded so different than the live club that I listened to it twice, both sides. What I had on tape was the full-range signal that went to the active crossover, amps, and loudspeakers. It was apparent that the huge difference in sound recorded vs. live was a result of the loudspeakers.



Tom Danley is pictured during a payload delivery at NASA's Marshall Space Flight Center (MSFC), Huntsville, AL.

I wouldn't have guessed it at the time, but each of these events sparked an interest in sound, electricity, and loudspeakers or seemed to have set my compass heading.

SHANNON: What was your first audio technology project? Can you describe it? Is it still in use today?

TOM: The first audio thing I was ever paid for was a pair of speakers I made for my grandfather. When I was about 17 or 18, he commissioned me to build a pair of speakers for his basement. I made many pairs of loudspeakers and a couple sound systems in the 1970s, but I doubt any of them still exist. My first successful commercial loudspeaker product was in the 1980s.

During an airplane trip, I was paging through a surplus catalog when I thought of trying to use a surplus low-inertia DC servomotor to replace the voice coil in a woofer. One motor caught my eye, and I had seen two of them at the local surplus store, Harrison Supply. When I got home, I went and bought them for \$12 each.

My third prototype sounded good enough to bring to work and demonstrate. My bosses were both impressed, and Roy, who was a hi-fi buff and World War II acoustician from Mullard Labs, later told me I could start a small loudspeaker division in the shop as long as its only cost was space and lights. I had some "cred" by then with Roy, who knew I had hand built electrostatic speakers, and he asked me to disable the protection spark gaps in his ESL-63s. (They sparked at just the wrong times.)

Very hesitantly I agreed and took them home to take apart and "fix." The servodrive woofers we made there were used for special effects at Disney World, the Mirage casino's volcano, and the "low end" for tours for Michael Jackson, U2, Garth Brooks, Def Leopard, and so on. While these were much more powerful than conventional subwoofers, the motors were remnants of the computer tape memory era, and when they were no longer needed, the price for the motors skyrocketed them into extinction.

SHANNON: Tell us about your company, Danley Sound Labs, and the role you play within the company. To what do you attribute your company's continuing success?

TOM: I credit my associate Mike Hedden (Danley Sound Labs president/CEO) with much of the company's growth. And he followed a very logical but nontraditional path to expand the company. I am blessed to have him as my co-worker and friend. As for

my part, I invented and developed the technologies we use, including the Synergy horn, the Tapped horn, the Paraline, the Shaded Amplitude horn, and the Layered combiner. I also designed most of the products that use them.

Actually those things are a small part of what a company does. A good idea on its own is something like a one-legged table (not that useful on its own). While my job is the most fun (I think), when I send my drawings to Tom Wilson, our CAD guy, he figures out the easiest/best ways to make the product. After a final look, his 3-D CAD drawings are sent to the cabinet shop where the CNC machines get programmed based on those drawings and so on. Early on, I may make a prototype or mock-up a part of it to test, but the very large cabinets are too complicated to build at home.

Our success? Again, it is a number of things acting together, a synergy. The team Mike has assembled including Ivan Beaver, Chad Edwardson, Tom Wilson, and Doug Jones, and so on are people who are passionate

about sound. Some years ago for a trade show, we had T-shirts made that said: "Sit down and listen. It's about the sound, it's why you got into audio."

For my part, the team members have more of an idea what they need to do than I do. When they tell me what they need, it is more an outline of what a product needs to do. The "how and what" part is entirely up to me. I like to say that the better I see the target, the more likely I will be able design something to hit it. The best part is no one says you have to do it this way or that way. I can design and test whatever I imagine, as long as it gets the job done.

Sonically, Synergy horn's success is simple but one has to hear it. It is a way to combine more than one driver in more than one frequency range into a single, constant directivity horn with the appropriate crossover. The outcome literally appears to be, measures like, and sounds like a single crossover-less driver.

That sounds simple, but it is surprisingly hard to do. An example of

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Tom Danley's acoustic levitation research was featured on *Science* magazine's February 2000 cover.

its effect is that you can walk up to the speaker, move around, literally even put your head deep into the horn mouth of a three-way Synergy horn, and never hear any clue there is more than one driver. Many of them can even reproduce a square wave over a broad band, even with a passive crossover. In large-scale sound, this sounds dramatically different than the concert-style arrays.

SHANNON: You hold 17 patents on a variety of acoustic and electromagnetic devices. Where do your ideas originate?

TOM: Well, there are a couple new ones to add. To be current, the Tapped horn, Synergy horn, Paraline, and Shaded Amplitude horn have been issued, and the Layered combiner will be soon.

As to where they come from, I can try to describe the process. It begins with a problem to solve. The best solutions come when you find the simplest answer whose form is driven by the problem and conditions. With the Synergy horn, for example, the solution should have constant directivity, and it should act like one source in time and space so it will not have lobes and nulls in its dispersion pattern.

Sometimes solutions to these issues happen in a flash, sometimes it takes

months to work out a solution. The Layered combiner used in the J-series horns was the hardest loudspeaker design I have worked on so far. It took four months to get from a fleeting mental image of "I think I see how" to a drawing that could be built.

SHANNON: Can you describe one or two of your inventions?

TOM: Thirty years later, I still get a kick out of acoustic levitation. Using high intensity sound you can position/levitate an object in midair. In the 1990s at Intersonics, I demonstrated acoustic levitation for a movie called "Mystery of the Sphinx." If you visit www.youtube.com/watch?v=NhDmCuffwOI, the scene is about 50 min into movie.

If you Google "Argonne acoustic levitation," you can find a link (www.youtube.com/watch?v=669AcEB) to a recent video demonstration using the levitation sound sources I developed.

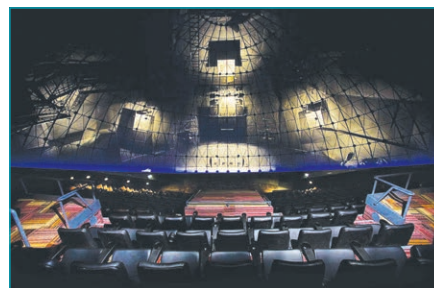
My inventions include the Synergy and Tapped Horn systems, and Shaded Amplitude Synergy horns. The Synergy and Tapped horn system is used at some of the large OMNIMAX to IMAX conversion theaters.

The Shaded Amplitude horns are the "things" that look like birdhouses that overhang the edge of the balconies at Turner Field in Atlanta, GA. With the Shaded Amplitude horns, energy projected to the front row is much louder than that projected directly below, so the measured sound pressure level (SPL) only varies about ± 1 dB over the seating area.

SHANNON: Which of your inventions would you describe as a "game changer" in your field?

TOM: My "game changer" would have to be the large J-series Synergy horns. In the stadium sound area (e.g., football- and baseball-sized venues) it is most difficult to get good sound.

It is also an area where the decision makers don't care about brand identity or who tours with X, Y, or Z speakers. There are no live-sound style riders and basically the only thing they care about is getting better sound. So for



Tom Danley's Synergy and Tapped Horn loudspeaker system design can be found in the Chicago's Museum of Science and Industry's IMAX theater.



Shaded-amplitude Synergy horns are shown overhanging the balcony at Atlanta's Turner Field.

a company like ours that doesn't do a lot of marketing, this is a level playing field. The concert systems that evolved for that market may be the best they know how to make, but they radiate an interference pattern you can hear if the wind blows. Every location has a different sound and radiate sound in the wrong directions. When you can eliminate all that interference, radiates enough acoustic power with one single source, *and* have more directivity, then it becomes a pretty obvious sonic choice. It's as close to hi-fi as you can get on a grand scale.

The familiar large sound systems are a huge array of sources, which depend on DSP, and sound worse the larger they are. The Synergy horns work because the source is acoustically correct at the origin.

SHANNON: Do you have any inventions that might soon be released?

TOM: Oh, now that wouldn't be fair. But, at InfoComm 2013 (in June), we introduced a new stadium-sized Synergy loudspeaker with a working distance of 1,000' that utilizes more than 100

drivers acting as one source.

SHANNON: Where does your affinity for low frequencies originate? What makes them so appealing to you?

TOM: At about age 9, I remember helping my grandfather clean up after a service at our church. I used to explore the building and on a previous visit I found a ladder in a back closet that led to the organ pipe loft where I found a confusing maze of tubes and pipes. Very complicated plus wires plus sound equals very interesting to me. I was especially interested in the device after hearing about a time when the organ's blower motor failed. The organist banged on the floor and my grandfather and two other church members managed to turn the shaft with their feet enough for her to play weakly. As a child, I loved fans so this was already cool.

That day, I spied the organist coming back to practice and, being curious, I snuck into the pipe loft. In that room, the pipes—rather the sound—

shook my body. I didn't know if I should run or stay perfectly still. Nothing was moving or otherwise scary, there was just the intense sound, so I stayed and listened for a while. It was by far the loudest musical experience of my life for a good number of years.

SHANNON: Where do you see the audio industry in 10 years?

TOM: In a way, the direction some of the industry has gone is disappointing. I would have given a great deal to be able to record like you can now with even a MI grade recorder, yet there are fewer bands. And, there are even fewer bands that can play well live or even have people who are interested in playing instruments. Like many industries, the home audio market is so focused on profits that fantastic technology goes unused, and many companies have lost focus on the need for high-quality audio products. Great progress exists for measuring and DSP products but often the companies are more focused on profits than

the advancement of the art.

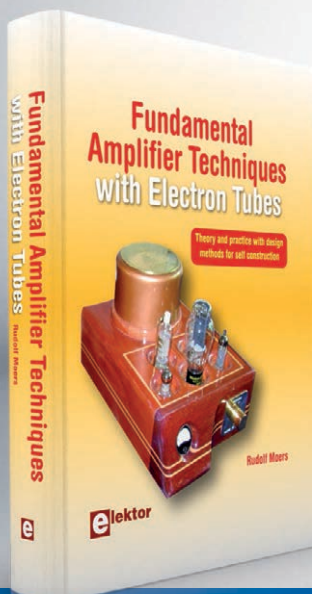
In the late 1990s, I brought a state-of-the-art TDS measuring system that was the size of a suitcase and the weight of a car battery to Egypt to measure the Great Pyramids' acoustics for a movie (www.livesoundint.com/archives/2000/julyaug/pyramid/pyramid.php). It was a serious armload to carry. Now, you can preform many more measurements and do them faster using an iPad and an interface.

As for me, I would say we still have a ways to go before you can be reliably fooled by a reproduced sonic "environment" or even being able to capture a remote environment. If you include a loudspeaker in a simple "old-time" generation loss recording test, even an anechoic one (where there is no room contribution), you find even good loudspeakers—unlike any other part of the chain—only go a few generations before sounding bad. I hope the industry eventually puts more emphasis on sound quality, the recording industry ends the volume wars, and the listeners win. Keep building. *aX*

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I had seen this type of misbehavior before, and had suspected that I was creating ground loops when probing those circuits. Previously, however, I had always been on a deadline and too preoccupied with debugging the circuit to stop and understand what was wrong with my probing technique. This time it was a weekend, and so I paused to study what was happening and understand why. It enabled me to understand how my oscilloscope probing was creating ground loops, and I learned a simple rule to improve my probing technique.

THE SETUP

Figure 1 shows the hardware I had on my bench that Sunday afternoon. Considering this simple setup, where should you clip your oscilloscope probe return before making measurements on this circuit?

The **Figure 1** test setup is fairly common. The device under test (DUT) is an amplifier, but it could be almost any electronic device. The DUT is powered by a bench-top power supply with an isolated output. Unplug the power supply and use a digital multi-meter (DMM) to ensure that the power supply output return is isolated from its power cord's third prong. This power supply could be either integrated into the amplifier or a separate bench supply (as it is in this case).

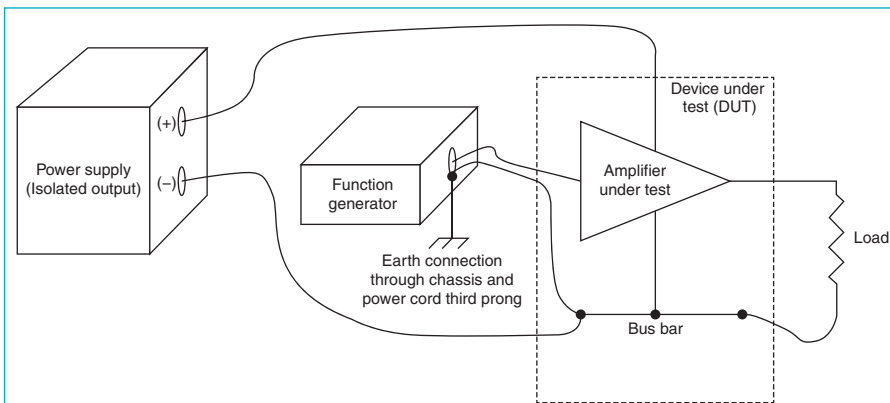


Figure 1: This shows a typical test setup for an electronic device.

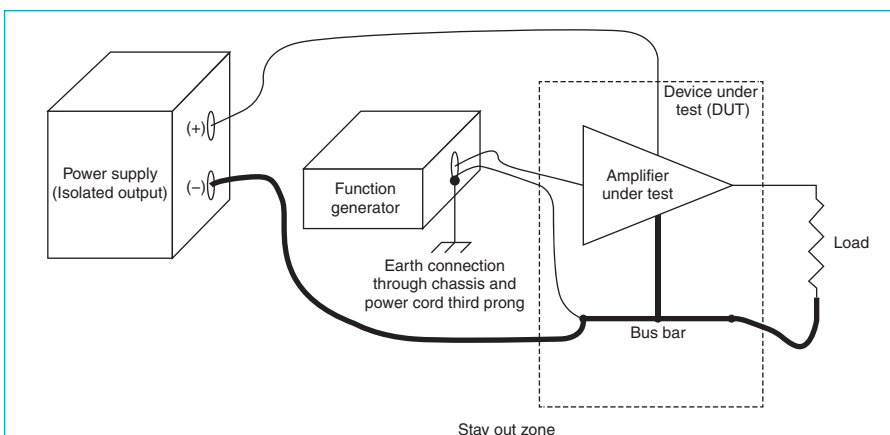


Figure 2: The “stay out zone” shows areas where you should not clip your oscilloscope probe return.

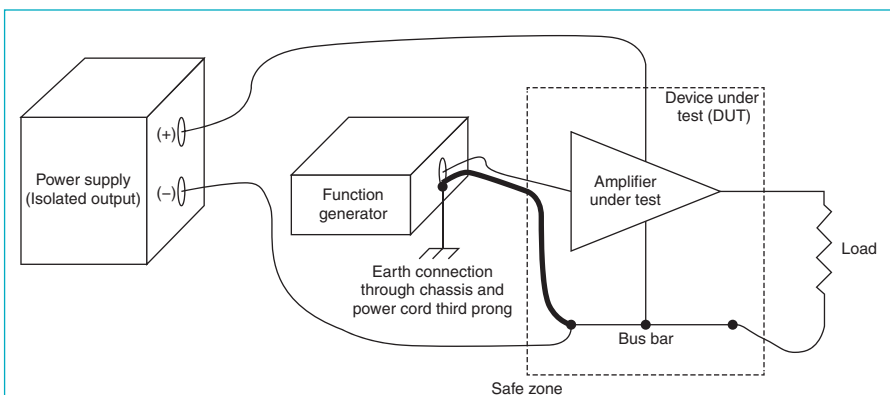
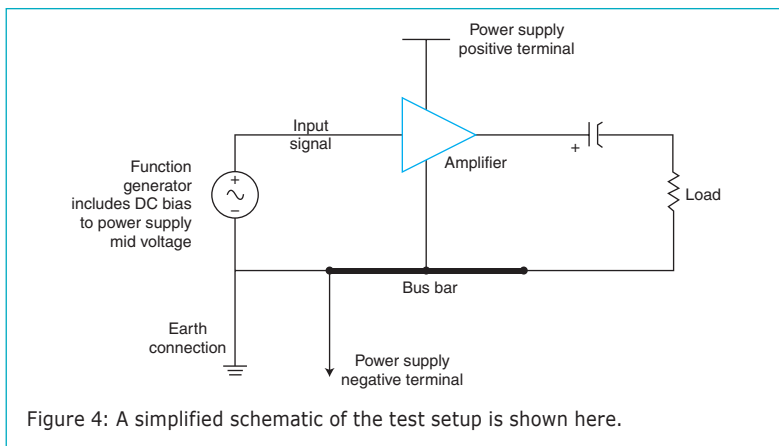


Figure 3: The “safe zone” is the area where you can clip your oscilloscope probe return without creating a ground loop.



The DUT is driving a ground-referenced resistor.

A function generator uses an earth-referenced output signal to command the DUT. The function generator's output signal return (often the shell of a BNC connector) is electrically connected to its chassis and the power cord's third prong. (You can unplug the function generator and check this with a DMM.)

The system's grounds are both proper and safe. There are no ground loops. The system has a single attachment to "Earth" at the function generator's signal return. Since the signal return dead ends at this connection, no current flows through that path. It simply provides a safety ground.

TEST SIGNAL MEASUREMENT

Suppose you have an oscilloscope and you want to measure signals on this circuit. Where should you connect your oscilloscope probe's ground clip?

My initial answer to this question was to clip my oscilloscope return to the bus bar on the DUT. I chose the op-amp's voltage reference—the point where the op-amp's return lead connects to the bus bar. This seemed like an obviously correct answer.

Interestingly, that is a wrong answer. Clipping on the bus bar (or anywhere in the return path through which load current is flowing) will create a serious ground loop. Load current, on its way to the power supply's negative terminal, will flow through the oscilloscope's probe return, down the oscilloscope power cord's ground wire, up the function generator power cord's ground wire, and back

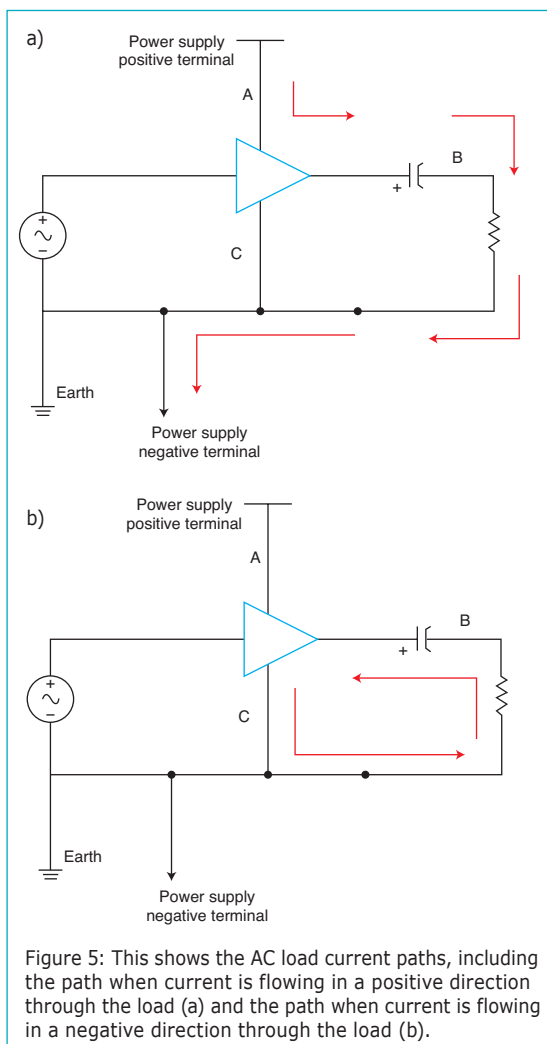
through the function generator's signal cable return. This ground loop will alter the circuit's current flow and disrupt its nominal operation.

Figure 2 shows the "stay out zone"—the area where you should not clip your oscilloscope probe return. If you reference your probe to this part of the return path, you will create a ground loop with load current flowing through your oscilloscope's probe return.

Conversely, **Figure 3** shows the "safe zone"—the area where you can clip your oscilloscope probe return without creating a ground loop.

THE PROBE RETURN

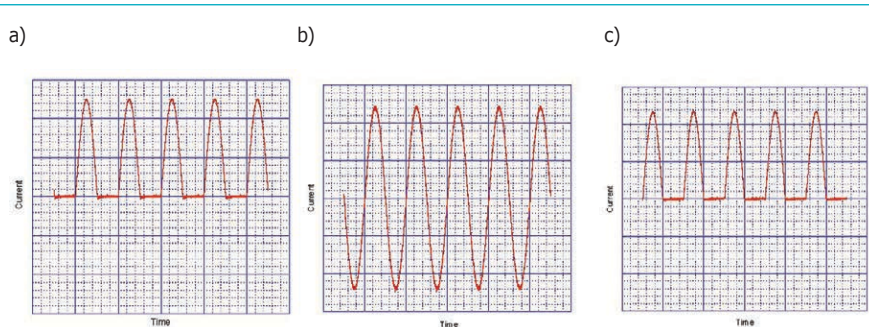
I have formulated an intuitive explanation of why the DUT's bus bar is not the correct place to clip your oscilloscope probe return, why nowhere in the "keep out zone" is safe, and—most



importantly—why the input signal return is the correct reference for your oscilloscope probe.

First, consider how load current flows through the circuit. **Figure 4** is a simplified schematic of the test setup.

Assume the function generator's AC input signal is biased to a voltage midpoint between the power supply's



positive and negative terminals. When the AC input voltage is positive (above the bias voltage), the current flows through the load following the current path shown in **Figure 5a**. When the AC input signal is negative (below the bias voltage), the current flows through the load following the current path shown in **Figure 5b**. Note that none of the load current flows to, or through, the earth connection.

Figure 6 shows the currents flowing through points A, B, and C on **Figure 5** (neglecting the amplifier's bias current) measured in an actual test circuit, positive currents are indicated by the arrow directions in **Figure 5a**. The currents shown in **Figure 5** and **Figure 6** demonstrate nominal circuit operation.

Now, consider bringing an oscilloscope onto the bench (see **Figure 7**). The oscilloscope's signal return, like the function generator's signal return, is electrically connected to earth. This means that when you attach the oscilloscope return clip to the circuit, you are electrically connecting that point directly to the function generator's return. Intuitively, that tells you the probe return should be clipped to the function generator return to avoid creating any new current paths—and this is correct.

Let's examine what happens when referencing an oscilloscope probe, as I did, to the point on the DUT's bus bar where the op-amp return lead is connected (see **Figure 8**).

Figure 9 shows what this means relative to the electrical schematic. The oscilloscope probe return has added a connection between the function generator and the DUT's bus bar. This connection is obviously going to shunt across part of the bus bar (from point "a" to point "b" in **Figure 9**).

When the amplifier drives current into the load as shown in **Figure 5a**, the current flows between point "a" and point "b" shown in **Figure 9**. Assuming the amplifier has low output impedance, a simple schematic of this load current path can be drawn (see **Figure 10**).

Figure 10 shows that part of the load current will flow through the oscilloscope probe, through the facility ground, and

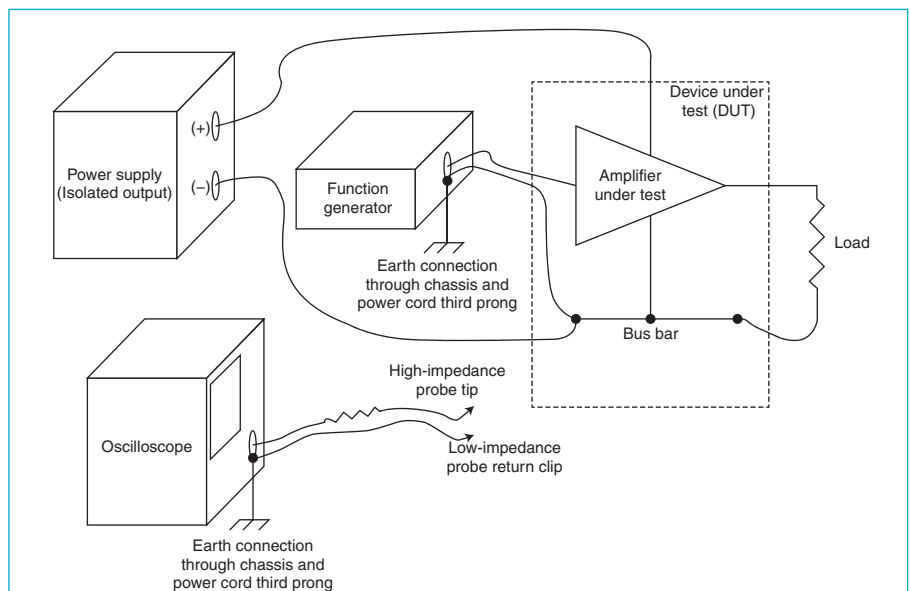


Figure 7: Bringing an oscilloscope onto the bench creates the potential to connect the function generator's input signal return to any point on the circuit through the probe's return clip.

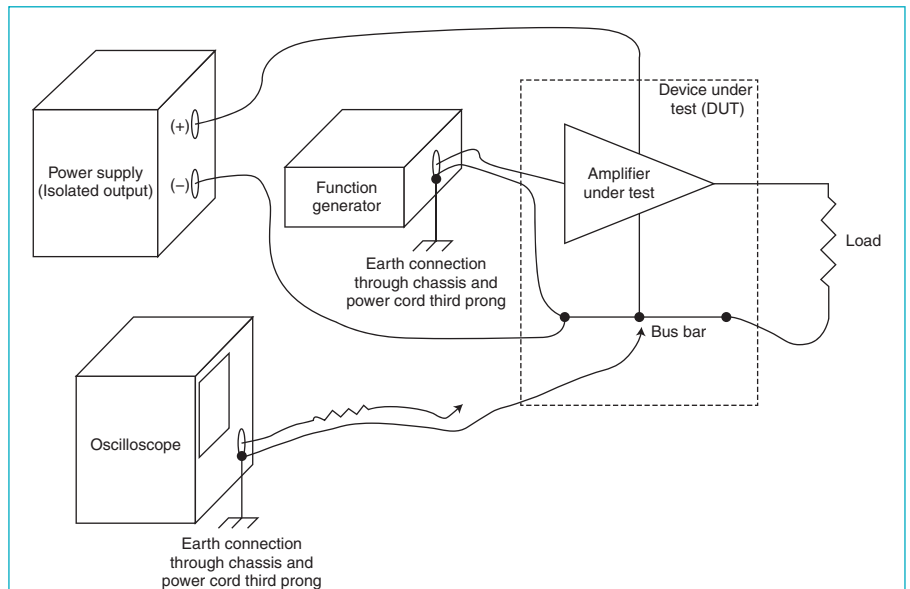


Figure 8: When referencing an oscilloscope probe to the DUT's bus bar, connect that point electrically to the function generator's signal return through both instruments' safety ground "Earth" connection.

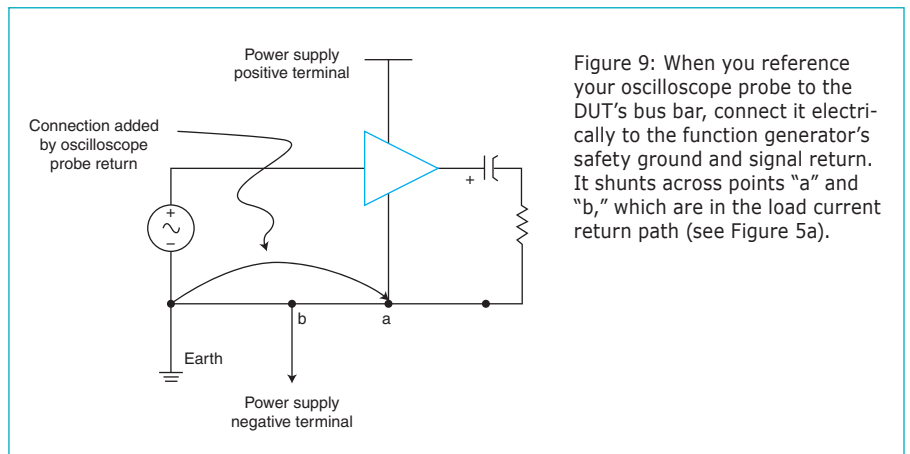


Figure 9: When you reference your oscilloscope probe to the DUT's bus bar, connect it electrically to the function generator's safety ground and signal return. It shunts across points "a" and "b," which are in the load current return path (see Figure 5a).

through the function generator signal return on its way to the power supply's negative terminal. **Figure 11** shows a measurement of current flowing

through the oscilloscope probe return. The graph is scaled and time-phased identically to the graphs shown in **Figure 5**. A significant per-

centage of the bus bar return current is flows through the oscilloscope probe return. This is a ground loop, "common mode" current that also flows through the function generator's signal cable return, causing signal noise.

Figure 10 can be redrawn to show where it is safe to clip your probe return—anywhere off the load current path—or equivalently, anywhere on the input signal return (see **Figure 12**).

GROUND LOOP

Figure 13 shows this ground loop current path in the hardware setup. It is easy to see how the load current, on its way

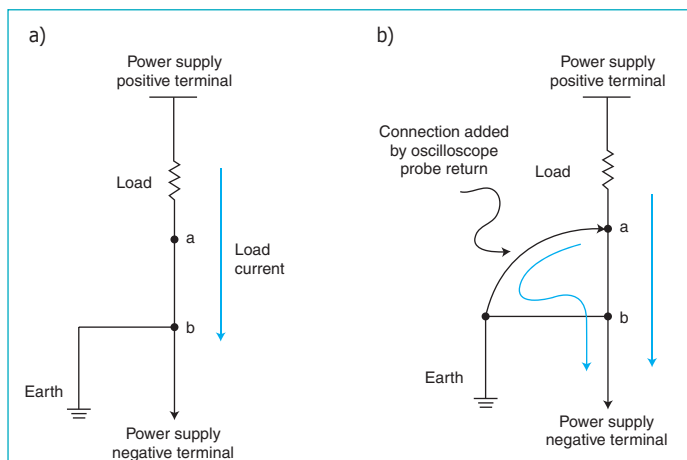


Figure 10: The load current should flow from the power supply's positive terminal directly to its negative terminal (a). When an oscilloscope probe return is clipped to the bus bar, part of the load current is diverted.

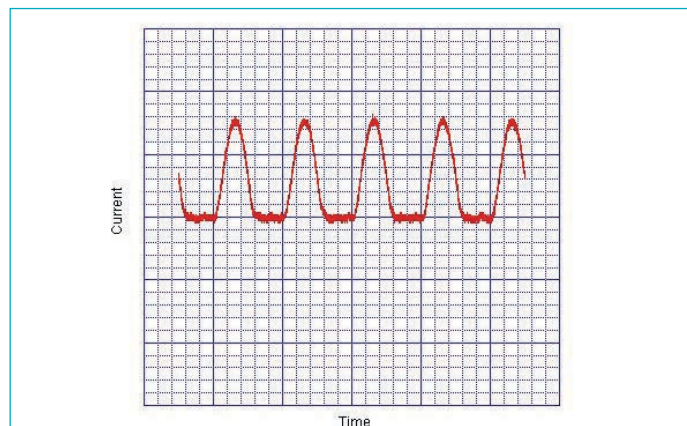


Figure 11: The ground loop current flows through the oscilloscope probe return (and the function generator signal return) when the oscilloscope probe return is clipped to the DUT's bus bar at the op-amp reference pin.

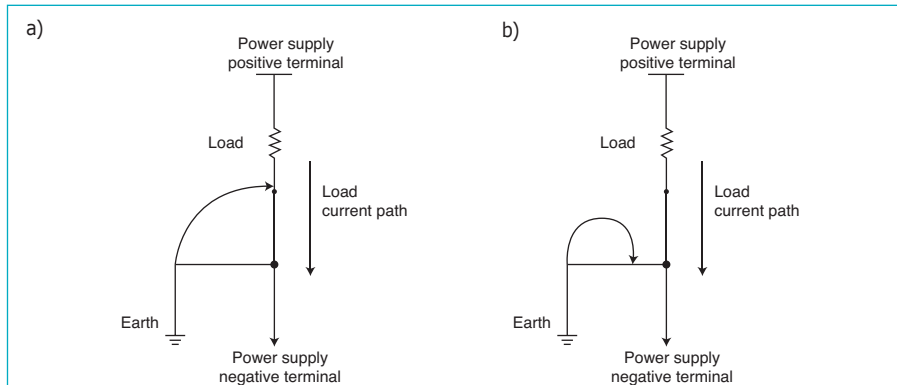


Figure 12: Don't clip your oscilloscope probe return anywhere in the load current path. This shorts across part of the current path and enables the ground loop current to flow through the oscilloscope and the function generator returns (a). Referencing your probe to any point on the input signal return path avoids creating ground loop currents (b).



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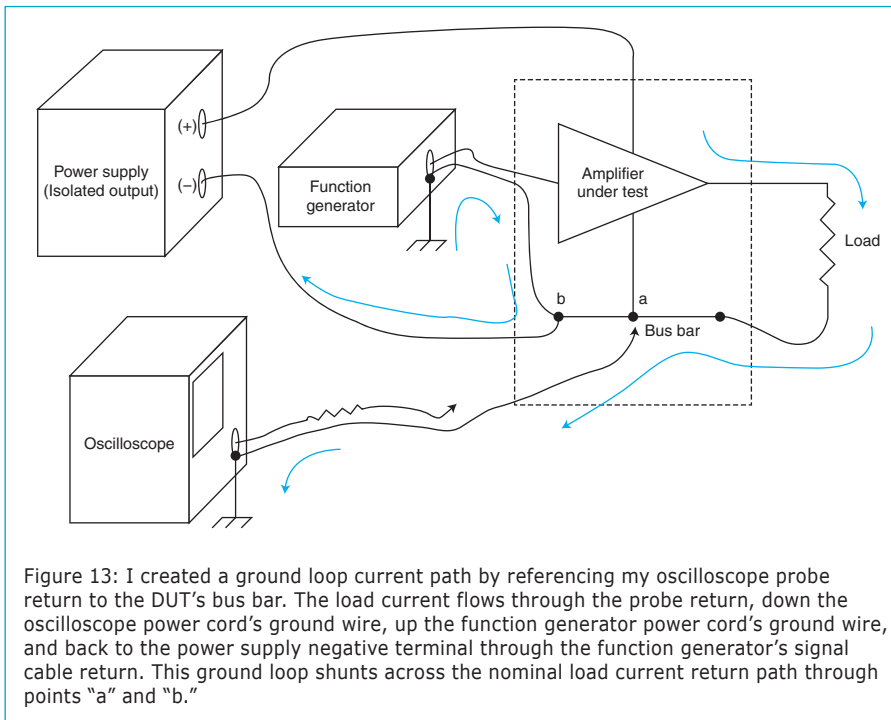
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back to the power supply's negative terminal, is sidetracked through the oscilloscope's probe return, down the oscilloscope power cord's ground wire, through the facility ground, up

the function generator power cord's ground wire, and back through the function generator's signal cable return.

Sometimes another technique is

recommended to avoid these ground loop currents. You can use two-prong "cheater plugs" to plug your test equipment power cords into the wall. These adapters break the connection between earth and the instrument chassis by disconnecting the power plug's "third prong." This also defeats the equipment's safety grounding.

If, due to a malfunction, the "hot" AC line (internal to the instrument) becomes electrically connected to the instrument chassis, you would be in danger of electrocution. A safer approach is to understand your current paths and avoid creating ground loops by using good probing technique.

To summarize, I learned one simple lesson when probing earth-referenced circuits—don't clip your probe return anywhere on the DUT's return current path. Instead, reference your oscilloscope probe to the existing earth-referenced point, which, in this case, is the input signal return. [aX](#)

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MEMBER PROFILE

Member Name: Fabio Camorani

Location: Forlì, Italy

Education: Electronic Engineering, University of Bologna, Italy

Occupation: Fabio is a lean production and audio expert for a big international company. He also has two small Italian audio brands and a very small top-quality vinyl label.

Member Status: Fabio said he doesn't recall the exact year he started subscribing to the Audio Amateur publications, but he said it was sometime during the 1990s. He added that he has all the *The Audio Amateur* and

Glass Audio issues, and now he subscribes to *audioXpress*.

Affiliations: He has been an Audio Engineering Society (AES) member since 2000.

Audio Interests: Fabio said he is interested in all audio aspects, from recording to speakers.

Most Recent Purchase: His most recent purchase was a Fidelity Research FR-1 MK3 phono cartridge.

Current Audio Projects: Fabio is working on a system that contains a 32-bit DAC and four power amplifiers, including one big output transformerless (OTL) amplifier, a phono preamp, a line preamplifier, and a super phantom power supply unit.



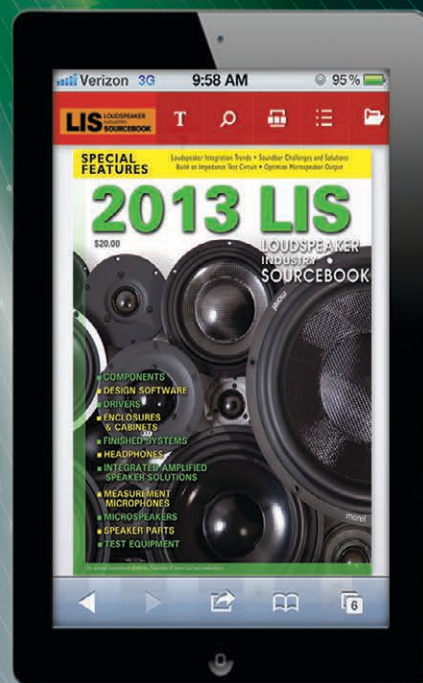
Fabio Camorani

Dream System: Fabio said his dream system is a four-way speaker system with Goto Berillium diaphragms and multiamplified compression drivers. The system would also have eight mono pure triode single-ended amps and an air-bearing turntable.

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Parasound's New Zdac

Reviewer praises Zdac's overall performance

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Price: \$475

Many *audioXpress* readers are already familiar with Parasound, an American audio company located in San Francisco, CA. Parasound has earned a reputation in high-end audio circles for offering products with excellent performance and reliability at fair prices. Its product line covers most of the spectrum, as far as cost is concerned.

The premium Halo line features several preamplifiers and power amplifiers designed by John Curl, one of the world's most respected high-end audio designers. The company's latest Halo line offering is the CD 1 compact disc player—a joint design effort with the Danish audio firm Holm Acoustics (www.holmacoustics.com). The design represents a radical departure from the conventional method of retrieving data from an optical disc. The CD 1 uses a CD-ROM drive and an Intel ITX computer to read a disc at high speed, buffering the data until an error-free retrieval has been achieved. (Holm Acoustics sells its own CD 1 player, which appears to use the same technology but has a completely different cosmetic design.)

Parasound's NewClassic line features four mid-priced power amplifiers and a matching preamplifier. The Z-series is its ultra-compact line of components, which includes a phono preamplifier, an FM/AM tuner, a two-channel zone preamplifier, and a matching power amplifier. I use a Zamp v.3 with my computer audio setup in my office and a NewClassic 2125 power amplifier in my production



Photo 1: The Parasound Zdac's front panel shows the AC power switch, headphone volume control and jack, input display, and input selector switch (a). A silver version, without rack mounting ears, is also available. The Zdac's rear panel contains balanced and unbalanced analog outputs, Toslink optical, S/PDIF coaxial, and USB digital inputs, along with an IEC power connector (b). (Photo courtesy of Parasound)

studio at The Crane School of Music, SUNY Potsdam. Both products provide excellent performance and reliability.

THE Zdac

The Zdac is Parasound's latest Z-series offering and its first digital-to-analog conversion (DAC) product in about 13 years (see **Photo 1**). Like the Wyred-4Sound μ DAC and the NuForce Icon HDP (which I reviewed in *audioXpress* February 2013 and April 2011, respectively), the Zdac includes a USB input, making it ideal for computer audio applications. But unlike those portable units, the Zdac is intended for more permanent installations (e.g., shelf or rack mounting).

The Zdac's USB input is designed to take advantage of a unique Windows 7 feature. The Windows volume control also changes the Zdac's analog outputs. With Windows XP and the Mac OS, the volume control only affects the volume internally. It cannot control an external DAC.

If a USB DAC is connected directly to a power amplifier, you will have little control over the playback level and will probably find the playback volume excessive, even with the Windows volume

control operated at a low range. You will need a preamplifier or passive analog volume control between the DAC and your power amplifier. With Windows 7, the Zdac can be directly connected to a power amplifier, eliminating the need for a preamplifier. The Windows 7's ability to control the Zdac's output level gives you full control over the playback level.

Unlike the NuForce Icon HDP and the more expensive Benchmark DAC-1 (which Chuck Hansen and I reviewed in *audioXpress* January 2009), the Zdac's main outputs are not affected by the front panel volume control. The volume control only affects the headphone level. No driver is needed for the Windows or Mac OS X OSes. Your computer should automatically recognize the Zdac when it's powered up and connected to a USB port. The unit also includes S/PDIF coaxial and Toslink optical digital inputs plus unbalanced RCA and balanced XLR outputs.

VOLUME CONTROL

Parasound's Bob MacDonald gave me a Zdac datasheet, which offers the following caveat regarding the use of Windows 7's OS to control volume: "For

Zdac MANUFACTURER SPECIFICATIONS LIST

Channel balance: <0.05 dB, 20 Hz–20 kHz	AC power requirement: 10 W 110–130 VAC 60 Hz 220–240 VAC 50 Hz Voltage is selected on the chassis bottom
Channel separation: >95 dB, 20 Hz–20 kHz	
DAC: Analog Devices AD1853	
Frequency response: 20 Hz–20 kHz, ± 0.2 dB	Audio output level: Unbalanced: 2.1 V _{RMS} Balanced: 4.2 V _{RMS}
Sample rate converter: Analog Devices AD1895	
Signal-to-noise ratio (SNR): 110 dB, A-weighted	Dimensions: Width: 9.5"—241 mm (black); 8.5"—220 mm (silver) Depth: 10"—254 mm Height, with feet is 2"—51 mm; without feet is 1.75"—45 mm
Supports incoming digital word lengths: 16–24 bit	
Supports incoming digital sampling frequencies: Optical and Coaxial: 8, 11.025, 16, 32, 44.1, 48, 88.2, 96, 176.4, and 192 kHz	Output impedance: Line out: 160 Ω , balanced or unbalanced Headphone out: 40 Ω
THD + Noise: <0.015%	Rack mount accessory (black Zdac only): Side-by-side mounting (SBS) bracket: purchased separately Zblank: Panel extender, purchased separately (Silver Zdacs cannot be rack mounted)
Upsampling frequency: 422 kHz/24 bit (all inputs)	
USB: 44.1, 48, 88.2, and 96 kHz	Weight: Net 5 lb (2.3 kg); shipping 8 lb (3.7 kg)
USB streaming controller: Texas Instruments TAS1020B	

the absolute best sound, we recommend setting all computer volume controls to their maximum and using an analog preamp to adjust volume. Using the volume control built into your computer's operating system can reduce sound quality. This is because a computer's volume control typically reduces the overall resolution of the audio signal because it is using fewer bits to represent the original signal. For customers who prefer the simplicity (and cost savings) of using a Zdac straight into a power amp without an analog preamp, they can still use the volume control of their computer. This will come at the cost of a slight sound quality reduction. In this case, it would be best to turn the gain on the amp down a bit so that the computer's volume is operated higher on the scale. For example it would be better to start with your computer's volume closer to 70% than 30%."

I find the Windows volume control a nuisance, especially since it doesn't sit on top of your application. You have to click on it in the system tray and then adjust it. When I'm doing digital audio editing, I find it much easier to just reach over and adjust a conventional volume control, which is why the NuForce and Benchmark DACs remain essential parts of my digital editing systems at home and work.

RECLOCKING AND UPSAMPLING

Like the CD 1 I mentioned earlier, the Zdac's design was accomplished in collaboration with Holm Acoustics. The Zdac's digital front-end is an AKM Semiconductor AK4113 audio receiver. The AK4113 includes six digital inputs, eliminating the need for external input switching. The low-jitter analog phase-locked loop (PLL) will accept inputs sampling frequencies from 8 to 216 kHz. The Zdac also contains an Analog Devices AD1895 sample rate converter and the AD1853 DAC.

Holms Acoustics also chose the AD1853 DAC for its CD 1 CD player. Parasound notes that all inputs in the Zdac are reclocked and upsampled to 422 kHz. This requires some clarification, since the AD1895 can't upsample to frequencies higher than 192 kHz.

Parasound is referring to the combination of the AD1895's upsampling and the oversampling that takes place in the AD1853. MacDonald said: "To get the widest bandwidth on the AD1895 in master mode (self-generating I²S from master-clock on output), a 27-MHz crystal is used. 27 MHz is the highest allowed master clock on the AD1895. The word clock on the output of the AD1895 is 1/256th of the master-clock so the outgoing sample rate is 105.46875 kHz. The AD1853 is then up-sampling by

a factor of four so the AD1853 output sample-rate is 421.875 kHz (422 kHz)."

Parasound has not implemented the Red Book de-emphasis specification in the Zdac (verified using my De-Emphasis

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Test CD, *audioXpress* February 2009). This is strange since the AK4113 input receiver and the AD1853 DAC support Red Book de-emphasis. This will only affect a small number of very early CDs, mostly discs manufactured by Denon in Japan in the 1980s.

The USB input employs a Texas Instruments (TI) TAS1020B USB streaming controller, which is compatible with the USB 1.1 specification and supports standard sampling frequencies up to 96 kHz. The datasheet mentioned earlier also contains some additional details about the USB technology used in the Zdac. Although the TAS1020B has become an industry standard for USB audio applications, Holm Acoustics found "significant shortcomings in TI's internal software, which results in greater jitter on its digital output." The datasheet goes on to state: "In a typical USB DAC the jitter at the TAS1020B output causes the AD1895 sample rate converter to trigger its polyphase filter coefficient generator. Altering the coefficients generates uncertainty in the numerical accuracy on the digital data output of the AD1895, even though the output clock is dead-stable. So the end effect is like jitter on the

analog output of the AD1853 DAC IC. The result: impaired sound quality."

The note explains that "Holm discovered undocumented parts of the TAS1020B and replaced TI's own software with proprietary software, which greatly reduces jitter on the TAS1020B digital output. This is accomplished through numerical optimization of its phase-locked loop (PLL), providing a superior basis for maximizing the synergy of the AD1895 and AD1853 to further reduce USB timing errors. This means the AD1895 plus AD1853 duo needs to do less 'work' in bringing down the jitter."

A pair of TI's LME49740 quad op-amps are in close proximity to the AD1853 DAC chip on the main PC board. Since the AD1853 has differential current outputs, it's reasonable to assume that the 49740 is used for differential current-to-voltage (I/V) conversion and analog filtering. The LME49740 is one of the high-performance audio chips that became part of the TI product line when the company acquired National Semiconductor. The specifications on this op-amp include a 20-V/ μ S slew rate, a 55-MHz gain-bandwidth product, a 2.5-nV/ $\sqrt{\text{Hz}}$

voltage noise, and a 0.00003% THD.

Two sections of this chip, in each channel, appear to directly drive the balanced outputs. The small analog output board has a LME49720, a dual op-amp with the same specifications as the 49740. This chip feeds the unbalanced RCA outputs. The entire analog signal path is DC-coupled, and Parasound has ensured that DC offsets are well controlled.

I measured an exceptionally low 0.2-mVDC at each of the unbalanced RCA outputs. The headphone amplifier is a TI TPA6120A2, an integrated circuit based on current-feedback technology with a 1,300-V/ μ s slew rate. This device can deliver 80 mW into a 600- Ω load with only 0.00014% THD. The signal-to-noise ratio (SNR) is 120 dB, making it suitable for 24-bit digital applications. This chip has differential inputs, enabling it to take full advantage of the fully-balanced I/V conversion and analog filtering provided by the LME49740.

LINEAR POWER SUPPLY

A linear power supply powers all of the Zdac's circuitry, and includes

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a hefty toroidal power transformer (see **Photo 2**). Separate regulation has been employed for the ± 9 -V analog supplies, and the 5- and 3.3-V digital supplies. Since portability was not an issue, Parasound was able to avoid the switching wall-wart supplies used with most portable USB DACs, including the NuForce Icon HDP and the Wyred4Sound μ DAC. The Zdac is supplied with an 18-AWG International Electrotechnical Commission (IEC)-style power cord, but this DAC is refined enough to warrant replacing it with a heavier gauge cable. I used a 2-m D.H. Labs Red Wave AC power cord, a

PERFORMANCE

The Zdac is sonically the best of the three under-\$500 USB DACs I've reviewed so far. Sonically, Parasound has clearly set its sights on more expensive, fixed-installation DACs rather than portable products, and its efforts have paid sonic dividends.

The most striking aspect of its performance is the large soundstage, which is both wider and deeper than its portable competitors. The Zdac also offers a neutral tonal balance, with a smooth, nonfatiguing treble.

In my review of the Benchmark DAC1 USB, I praised its exceptional inner detail and spacious soundstage, but I also noted a slight tendency toward brightness in the tonal balance. Although the Zdac doesn't quite match the Benchmark with regard to detail or soundstage size, it was nonetheless very impressive in these areas. I found the Zdac's tonal balance free of the Benchmark's treble brightness. Like the Benchmark, the Zdac aims at sonic accuracy rather than euphonic coloration, but it's a bit warmer and less analytical than the Benchmark. The bass region is extended and well-

I'm currently evaluating. The OPPO players offer stand-alone performance that surpasses much of the competition in their price categories. However, I would expect the Zdac to improve the performance of many other digital players in the \$500-\$1,500 price range.

In my Benchmark review, I also noted the processor's state-of-the-art jitter rejection. It seemed to eliminate the differences in the "sound" of various transports, differences that have been audible on every other D/A converter I've used. The Zdac did not quite equal the Benchmark in this regard.

Comparing the OPPO BDP-105 and my NAD M55 used as transports with an S/PDIF connection to the Zdac, the OPPO sounded smoother in the treble and delivered a larger soundstage than the NAD. But, the differences were not dramatic, and the NAD M55 still performed very well with the Zdac. The headphone amplifier is dynamic and punchy, and had no difficulty driving my AKG K701 headphones, which have a 62- Ω impedance rating. As implemented in the Zdac, the TI headphone

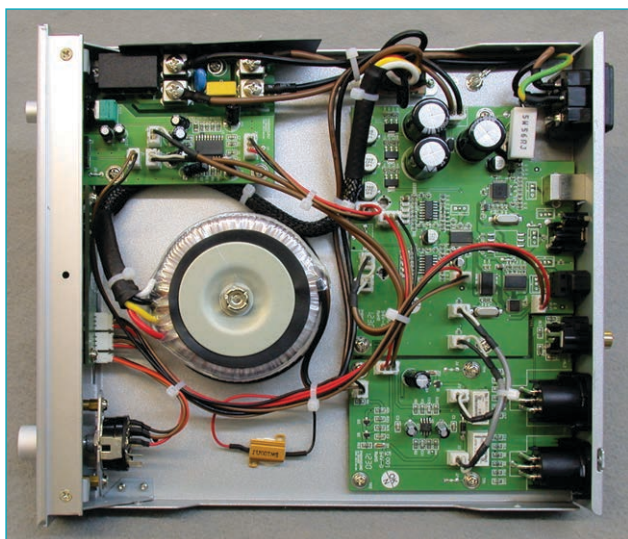


Photo 2: A large toroidal power transformer is at the heart the Zdac's linear power supply. The main PC board in the upper right contains all digital input circuitry—an Analog Devices's AD1895 sample rate converter and AD1853 DAC chip, analog op-amps, and a linear DC-supply regulation. The headphone amplifier board is in the upper left.

10-AWG double-shielded cable fitted with Wattgate gold-plated AC connectors (www.silversonic.com). This is admittedly overkill, but it did yield superb results. A cost-effective solution would be the Pangea AC-14, regular or signature version (www.audioadvisor.com), or D.H. Labs Power Plus bulk cable fitted with Marinco connectors from Parts Express (www.parts-express.com). Admittedly, I am rather skeptical regarding some of the claims made for exotic high-end AC power cords. But high-performance audio equipment will normally benefit from replacing 18-AWG cords with heavier-gauge wire and shielded cords can be beneficial, especially on digital equipment.

controlled, undoubtedly helped by Parasound's conservative linear power supply and DC coupling.

Although many buyers will employ the Zdac in computer or music server applications using its USB input, I think external D/A converters remain a popular way to upgrade the performance of stand-alone CD and multi-format players. I found that, sonically, the Zdac outperformed my NAD M55 player, offering superior detail, resolution, and soundstaging compared to the M55 as a stand-alone unit. But in these areas, it fell a bit short of my OPPO BDP-95 Universal Blu-ray player's performance (reviewed in *audioXpress* January 2012) and the BDP-105 that

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amplifier chip is clean, detailed, and spacious.

THE BOTTOM LINE

The three under-\$500 DACs reviewed in *audioXpress* each fill an important niche. The NuForce Icon HDP and Wyred4Sound μ DAC offer fine performance in portable units that will fit inside many laptop computer cases. The NuForce contains a headphone amplifier plus a preamplifier and volume control, making it ideal where an analog level control is needed.

Sonically, I found Wyred4Sound μ DAC even better than the NuForce unit. It will be an obvious choice where portability is needed but an analog volume control and headphone jack are not. (Wyred4Sound has recently introduced the μ DAC-HD, which includes a headphone amp, but the level must be set using the Windows volume control.) At \$399, the μ DAC is also the most affordable of the three.

From a purely sonic standpoint, the Parasound Zdac is free of the performance compromises that are inevitable

where portability is required. In my view, it is clearly the winner. If I could change one thing in the Zdac, I would prefer that the volume control would affect the main outputs and not just the headphone amplifier. This would eliminate the need for a preamplifier between the Zdac and the power amp, and it would make it a low-cost, fully functional alternative to the Benchmark DAC-1 USB.

The Parasound Zdac is an excellent product. I think it has raised the bar for affordable digital-to-analog converters. *ax*

SOURCES

K701 Headphones

AKG | <http://us.akg.com>

AK4113 Audio receiver

AKM Semiconductor, Inc. | www.akm.com

AD1895 Sample rate converter and AD1853 DAC

Analog Devices, Inc. | www.analog.com

DAC-1

Benchmark Media Systems, Inc. |

www.benchmarkmedia.com

CD 1 player

Holm Acoustics | www.holmacoustics.com

Intel ITX computer

Intel Corp. | www.intel.com

M55 digital disc player

NAD Electronics | <http://nadelectronics.com>

Icon HDP

NuForce | www.nuforce.com

BDP-95 Universal Blu-ray player and BDP-105 Blu-ray player

OPPO Digital, Inc. | www.oppodigital.com

Z-series components

Parasound | www.parasound.com

TAS1020B USB streaming controller, LME49740 quad op-amp, and TPA6120A2 headphone amplifier

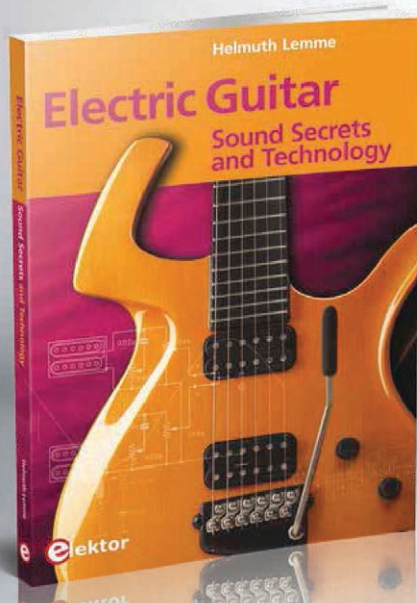
Texas Instruments, Inc. | www.ti.com

μ DAC and μ DAC-HD

Wyred4Sound | www.wyred4sound.com

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Output Transformers

Understanding output transformers helps in designing or tweaking audio power amplifiers



Hollow-state power amplifiers tend to produce high-voltage, low-current outputs. In other words, they are by nature high-output-impedance devices. Most speakers have low impedances—about 4–16 Ω . An output transformer is usually used to bridge this difference in impedances.

A number of designs have been created, mainly in recent years, to avoid the use of output transformers with their inevitable shortcomings. Perhaps the most common approach is to use multiple power tubes in parallel, connected in a cathode-follower arrangement. These designs are inherently

limited in output capability by the maximum plate or cathode current of the output tubes. For example, a triode-connected 6550 can handle about 175 mA. A fairly efficient home-stereo speaker will require about 0.6 A to reproduce the peak level heard from an orchestra playing at mezzo-forte (about 95 dB), if the listener is about 2 m from the speaker. This would require four 6550s in parallel. Alternatively, if the four 6550s were connected in push-pull, using an

output transformer, they could produce about 121 dB—if the speakers could handle it, and if the listener chose to risk his hearing. Thus, most listeners choose the transformer-coupled outputs.

THE IDEAL TRANSFORMER

An output transformer consists of a core of soft magnetic material with two wire windings. One winding—the primary—is connected to the output tubes, and the other—the secondary—is connected to the speakers. If a sine wave having an RMS voltage— V_p —is connected across the primary, the current will flow in the primary, generating an alternating magnetic field that will induce a corresponding sinusoidal voltage V_s across the secondary. The ratio of turns in the primary to turns in the secondary ($N_p:N_s$) determines the ratio of voltages ($V_p:V_s$) as shown in the equation below:

$$\frac{V_s}{V_p} = \frac{N_s}{N_p}$$

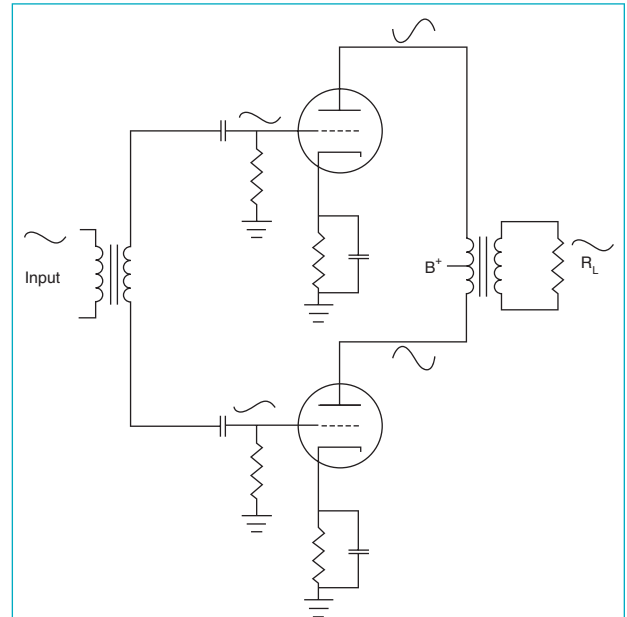


Figure 1: A transformer is shown connected in a push-pull output stage.

If there are 26 times as many turns in the primary as in the secondary, the secondary AC voltage will be 1/26 of the primary AC voltage. When voltage is stepped down, the current is stepped up by the same ratio, giving an impedance change equal to the square of the turns ratio. So a pair of output tubes operating into a transformer with a primary:secondary turns ratio of 26, and feeding an 8- Ω speaker will see a 5,408- Ω load (i.e., 8×26^2). This load impedance is usually specified in "ohms plate-to-plate." This means that since the transformer's primary is fed by the B+ supply, which is at AC ground, each tube sees a load impedance of:

$$\frac{5,408 \Omega}{2} = 2,704 \Omega$$

Figure 1 shows the push-pull triode output stage's schematic diagram. A slightly different circuit is often used with power pentodes and beam power

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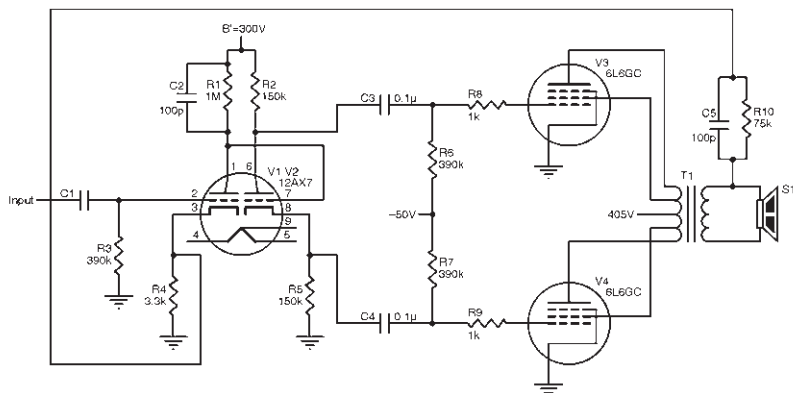


Figure 2: The ultralinear circuit applies negative feedback to the output tubes' screen grids, reducing distortion.

tubes. This circuit, called the "ultralinear circuit," feeds the power tubes' screen grids from special taps on the output transformer primary. By doing so, the ultralinear circuit applies negative feedback through the screen grids, reducing distortion in the output stage. **Figure 2** shows an ultralinear output stage.

When you specify an output transformer for a project, in addition to the impedances you must decide on the maximum plate current you will need for the tubes to draw. Selecting the proper primary impedance is beyond this column's scope. But you can look in a tube manual and find the examples listed for the output tubes you want to use, then specify the primary impedance given in the manual.

For example, if you want to use a pair of 6L6GCs to produce 55 W in a Class AB₂ configuration, you will need a 450-V plate supply and an output transformer capable of handling 210 mA of plate current plus 22 mA for the screen grid, and that presents a 5,600-Ω plate-to-plate load. You will also need to choose your desired secondary load values. Common secondary loads are 8 Ω (one speaker), 4 Ω (two speakers in parallel), 2 Ω (four speakers in parallel), and 70 V. The latter does not look like impedance, but it really is. It is the impedance into which the amplifier would produce a 70-V_{RMS} output when delivering maximum power. For a 55-W amplifier, a 70-V output winding corresponds to 89.09 Ω. (Why 70 V? Because the National Electrical Code specifies that circuits having a 70.7-V

maximum can be wired using low-voltage wiring techniques. High-voltage circuits must conform to several more regulations.)

I have just described the operation of an ideal transformer. If connected to a resistive load, such a transformer would present a simple resistive load to the output tubes. If connected to a complex-impedance load such as a speaker, it would present a higher-impedance version of that same load,

with the same phase angle, to the tubes.

Real transformers behave pretty much like ideal ones, but not exactly. **Figure 3** shows the equivalent circuit of a real transformer.

EQUIVALENT CURRENT

I will now discuss some of the effects represented in the equivalent circuit. The winding resistances cause conversion of electrical energy to heat, decreasing the transformer's efficiency. If too much heat is created, the insulation on the windings can be damaged, causing a short circuit between adjacent windings and ruining the transformer. Interwinding capacitances shunt energy across the windings at high frequencies, limiting high-frequency response. Core losses comprise both hysteresis loss and eddy-current loss. Hysteresis loss is energy that is consumed when the core's magnetization is reversed at the end of each half-cycle, so it increases with frequency. The choice of core material affects the amount of hysteresis loss. Eddy currents are circulating currents that flow in the



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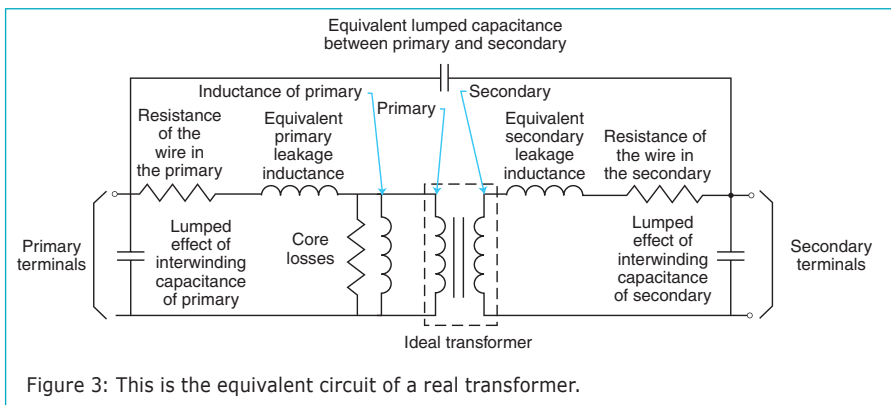


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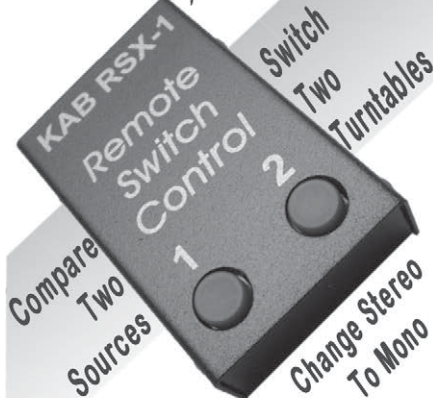
core (which is usually made of metal) because of the changing magnetic field created by the AC in the primary. Since the core has resistance, these currents cause core heating, which represents an energy loss from the transformer.

Leakage inductance represents the alternating magnetic field produced by the primary winding that does not link with the secondary winding. This magnetic field collapses back on the primary when the polarity of current in the primary is reversed, and the alternate storage and energy return from the leakage field constitutes an inductance.

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The ideal transformer's primary side acts like a reflected secondary impedance:

$$\text{load impedance} \times \left(\frac{N_p}{N_s} \right)^2$$

From the secondary terminals, it acts like an AC voltage source.

Transformers using larger gauges of wire in the windings will have lower winding resistances, making them more efficient and able to handle greater amounts of current. However, they will also have higher inter-winding capacitances and, thus, more limited high-frequency response. So you see that transformer design can become rather complex because of the interacting variables.

PERMEANCE

A given current in a certain number of turns produces a "magnetomotive force," (MMF, which is not really a force). The MMF produces a certain density of magnetic flux in a "magnetic circuit," depending on the ease with which the materials in the magnetic circuit can be magnetized. This characteristic of a magnetic path is called "permeance," and permeance is analogous to conductance (inverse of resistance) in an electrical circuit. For example, a coil of 100 turns of wire will create a much greater flux density if wrapped around the straight portion of a D-shaped piece of iron than if the coil were floating in midair, because the iron path has higher permeance than air. Permeance is a characteristic of a particular magnetic path—a given piece, or group of pieces of materials. Each type of material has a characteristic called "permeability," which relates to its ability to be magnetized. So pure annealed iron has a particular permeability, and a specific D-shape of

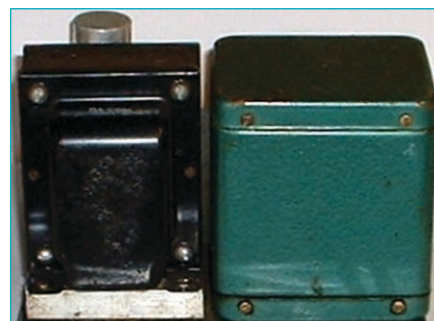


Photo 1: This shows a 1950s-vintage potted output transformer (right) and a 1970s non-potted one (left).

a specific size of pure annealed iron, has a particular permeance that depends on the permeability of pure iron and also the dimensions and shape of that particular piece of iron.

Permeability, and thus permeance, are not constant but they are affected by the strength of the magnetic field in which the material is immersed. Every magnetic material has a maximum flux density that it can support. Even if the magnetic material is surrounded by a winding having an increasing current, the magnetic field will not increase beyond a certain flux density. This effect is called "magnetic saturation."

When a transformer is magnetically saturated, it will generate significant distortion. The main instances of transformer saturation occur in single-ended (Class-A) amplifiers, which have a direct current in the primary all the time. In push-pull amplifiers, the MMF produced by the AC in the primary is basically canceled by the antiphase MMF produced by the AC in the secondary. One effect of the leakage inductance and winding capacitances is to shift the secondary AC's phase, so complete cancellation does not occur. An additional source of phase shift in the secondary current is nonresistive loads such as speakers. Using proper design methods, high-fidelity output transformers can be designed that have less than 0.05% distortion at 50 Hz, while providing design-maximum power output.

In years past, many output transformers were enclosed in rectangular steel cases and were potted in a tar-like substance to conduct heat away from the windings. **Photo 1** shows such a transformer alongside a more modern one.

Next month, we'll look at tone control circuits. *ax*

Beston's Horn-Loaded Transducer

A high-frequency, closed-back ribbon tweeter is measured for quality



Photo 1: Beston Technology's RT004A ribbon tweeter

This month, I received Beston Technology's RT004A, a high-frequency horn-loaded ribbon tweeter (see **Photo 1**). Beston Technology was founded in 2002 in Taipei, Taiwan. Since its inception, the company has focused on high-frequency ribbon devices using a Kapton substrate.

The Beston RT004A is a closed-back ribbon tweeter with two 7-mm × 112-mm apertures. Its features include a thin laminated aluminum diaphragm, an injection-molded horn-loaded faceplate, a stamped metal chassis, a neodymium (Nd-Fe-B) magnet, with a 60-W power handling rating (using a third-order at 1.5-kHz network), and a pair of solderable terminals.

I began analysis by performing a 300-point impedance curve (see **Figure 1**). As is common with ribbon devices, this transducer has only a slightly defined 1.68-kHz resonance. Minimum impedance for this driver is 7.9 Ω at 11 kHz with a just slightly lower 7.7-Ω direct current resistance (DCR).

Next, I mounted the RT004A in an enclosure with a 8" × 15" baffle area and measured the on- and off-axis sound pressure level (SPL). I set up the LMS analyzer to produce a 100-point 2.83-V/1-m gated sine wave sweep from 300 Hz to 40 kHz. I tracked the data in the horizontal and vertical planes with 0°, 15°, 30°, and 45° sweeps. **Figure 2** shows the Beston RT004A's on-axis response. The RT004A's frequency response is ±2.2 from 1.9 to 11 kHz, with the driver's second-order low-pass roll-off starting at about 12 kHz. Beston specifies sensitivity at 102 dB, and the SPL varies between 101–105 dB in the operating range.

Figure 3 shows the on- and off-axis frequency response in the horizontal plane, revealing the constant directivity radiation pattern. Since the aspect ratio of the RT004A's aperture is high, as are the majority of ribbon transducers,

there is substantial directivity in the vertical plane, as seen in the vertical on- and off-axis plot (see **Figure 4**). The two-sample, SPL comparison is shown in **Figure 5** indicating

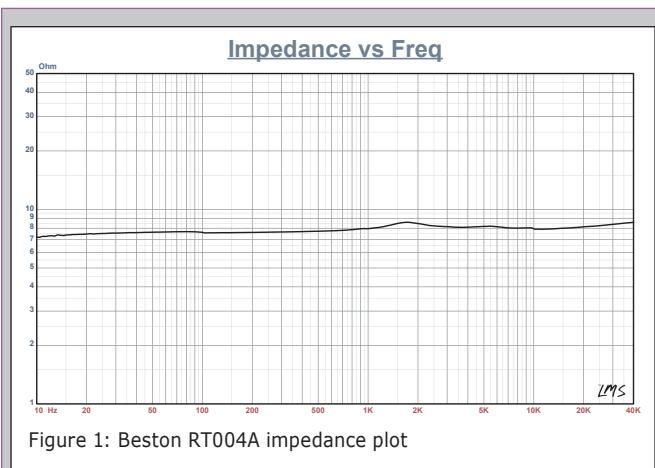


Figure 1: Beston RT004A impedance plot

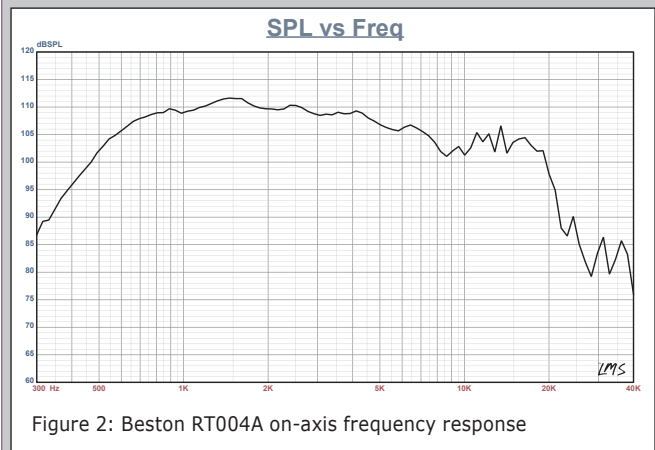


Figure 2: Beston RT004A on-axis frequency response

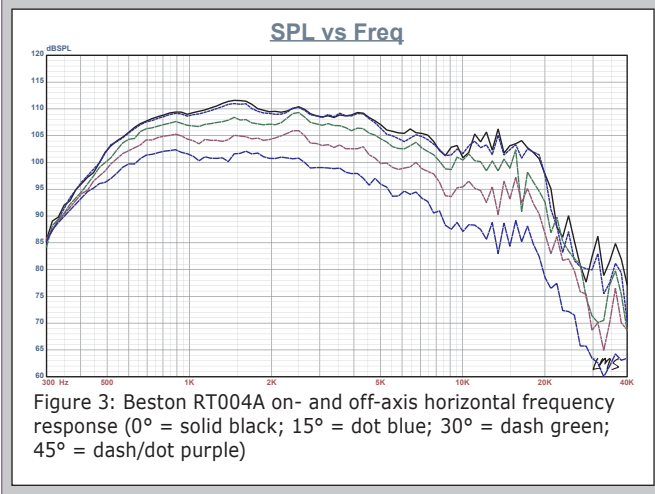


Figure 3: Beston RT004A on- and off-axis horizontal frequency response (0° = solid black; 15° = dot blue; 30° = dash green; 45° = dash/dot purple)

Looking for even more loudspeaker design inspiration?

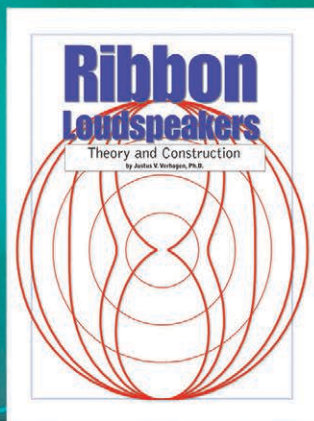
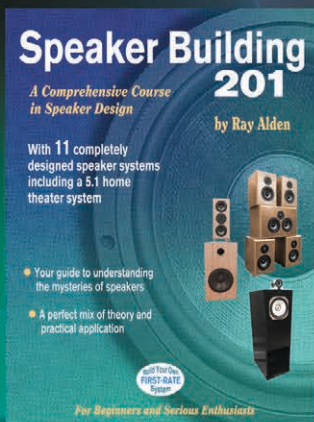
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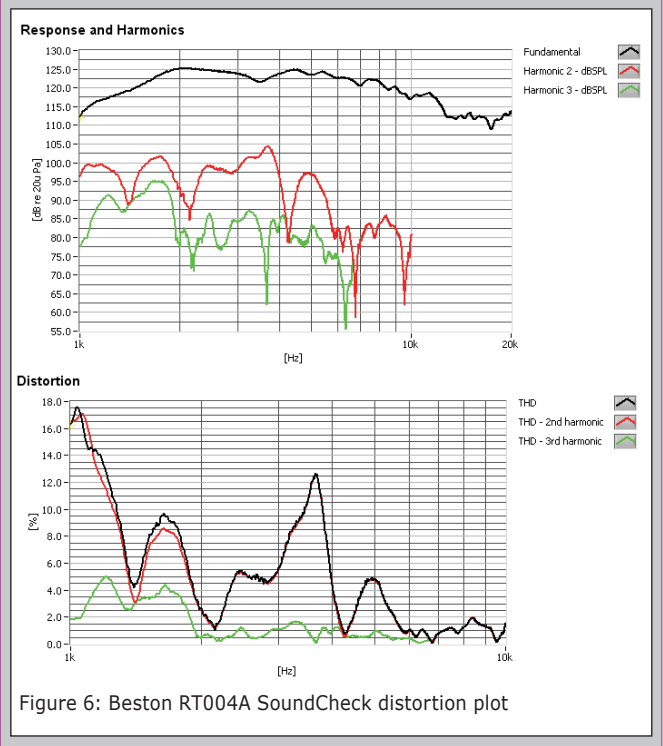
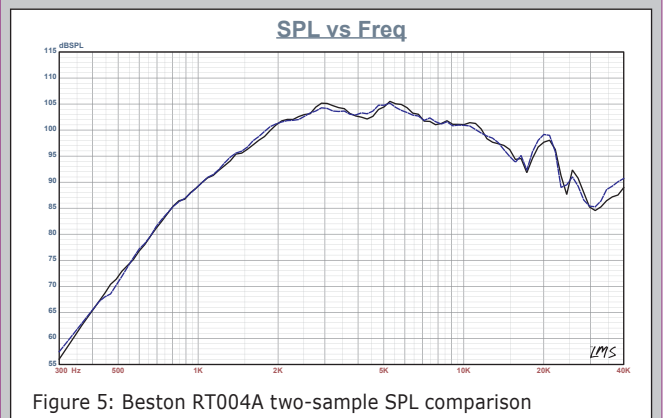
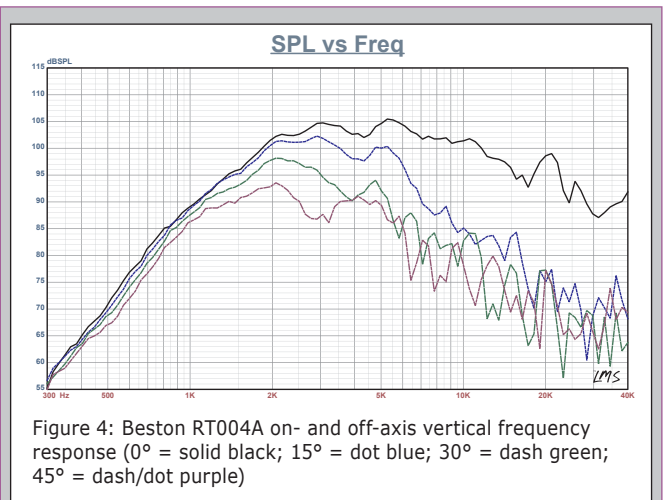
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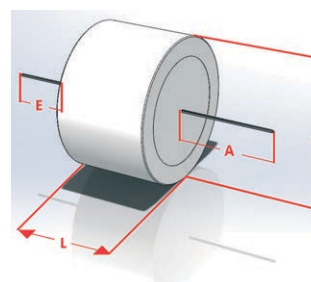
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that the deviation between the two samples was 1 dB or less throughout this driver's operating range.

For the last group of measurements, I set up Listen's SoundCheck SoundConnect analyzer (using SoundCheck 11



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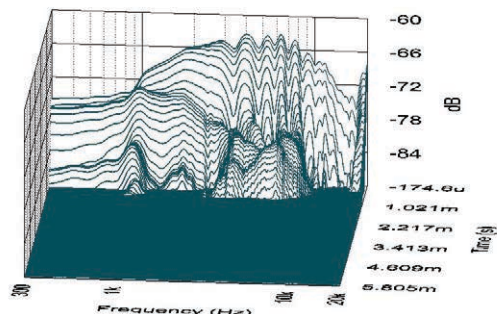


Figure 7: Beston RT004A SoundMap CSD graph

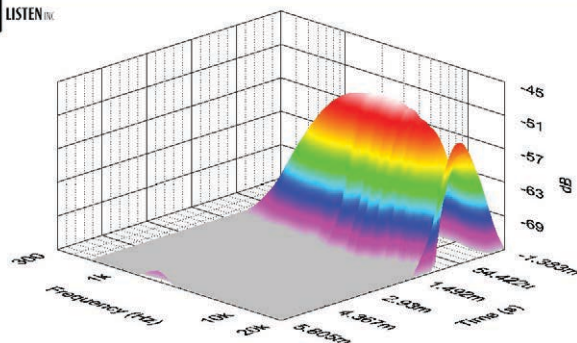


Figure 8: Beston RT004A SoundMap STFT plot

software—more about this next month) with the SCM microphone and SoundConnect preamp/power supply. I used the built-in pink noise generator and SLM utilities to set the SPL to 104 dB/1 m (4.2 V). Then, I relocated the 0.25" SCM microphone to 10 cm from the Beston RT004A's faceplate and ran the distortion curves shown in **Figure 6**.

For the last Beston RT004A ribbon tweeter measurement, I performed an impulse measurement and imported it into the Listen SoundMap software. I windowed out the room reflections and created the cumulative spectral decay (CSD) plot and the short-term Fourier transform (STFT), which are shown in **Figure 7** and **Figure 8**, respectively. For more information on Beston's high-frequency horn-loaded ribbon tweeter, visit www.ribbonspeaker.com.tw. *aX*

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Eminence Speaker recently introduced the D-fend SA300, a fully programmable stand-alone unit designed to protect passive loudspeakers from excessive power conditions. With patent-pending technology, D-fend enables maximum driver performance while ensuring damage-free operation.



The D-fend SA300 protects passive loudspeakers from excessive power conditions.

D-fend eliminates worries about blown speakers, HF drivers, or crossovers; or even worse, fire caused by excessive heat. D-fend keeps your system safe as well as your venue and audience. You simply set the thresholds and D-fend monitors and limits the amount of input power it passes through to the loudspeaker. It's USB compatible and can be programmed to your specifications from a desktop or laptop.

Operating from a standard speaker-level signal, the D-fend SA300 requires no auxiliary power unless it is used in low-power applications. D-fend loudspeaker protection is ideal for system installers, PA gear rental companies, OEM manufacturers, and end-users who own passive loudspeakers.

The D-fend SA300 is available to resellers through Eminence's dealers and distributor networks, and direct to consumers through www.D-fend.net. For more information, visit www.D-fend.net and www.Eminence.com/d-fend.

CUSH HEADPHONES RECEIVE AN UPGRADE

KICKER has added more features and conveniences to its Cush Talk headphones. KICKER Cush Talk headphones now provide a lightweight microphone and single multifunction button for

listening convenience and to easily transfer sound during phone calls. Cush Talk headphones also come with a protective storage pouch.

Featuring an ultra-lightweight design and thick over-the-ear cushions, Cush headphones provide comfort, detailed acoustics, a lavish fit, and now a microphone. With 54-mm speakers and a 118-dB maximum output, these headphones provide the bass response and tonal accuracy for which KICKER is known.

Cush Talk utilizes a 53", Kevlar-reinforced, flat cable to provide more freedom for movement. The flat cable is smooth and less prone to tangles. Kevlar, the same material used in military and law enforcement bulletproof vests,



KICKER Cush Talk headphones now provide a lightweight microphone and single multifunction button.

enables the KICKER Ultra-Gauge cable to perform under the most strenuous factory testing. The angled "L" plug (0.125", corrosion-resistant) provides strain relief and connects to any iPod, iPhone, MP3 player, or KICKER Docking System. For more information, visit www.kicker.com.

TRIAD SPEAKERS OFFERS A SPACE-SAVING SUBWOOFER

Triad Speakers, a leading custom manufacturer of home reference-quality loudspeakers, now offers the InWall Bronze/4 SlimSub. Measuring 3.938" deep to fit into a standard 4" deep wall cutout, the SlimSub's flat grill, flush-mount design represents an aesthetic improvement over the InWall Bronze/4 Sub, which protruded several inches from the wall.

The SlimSub delivers deep bass at high volumes, achieving a 109-dB



The SlimSub 10" aluminum driver is encased in a narrow enclosure with a flat flush-mount grill.

maximum output from 40–80 Hz with –6-dB bass extension at 25 Hz. This performance rivals the company's larger, 6" deep InWall Bronze/6 model.

The SlimSub fits into the same cutout (size "V") as the company's other in-wall subwoofers, making it ideal for retrofit opportunities. To blend with a home's décor, the SlimSub incorporates all three types of Triad's Acoustimesh grill (wide, narrow, and frameless), which the company can custom paint-match at the factory.

The InWall Bronze/4 SlimSub is paired with Triad's 350-W rack amplifier and costs \$1,400. Dealers can contact Triad Speakers for more information at 800-666-6316 or visit www.triadspeakers.com.

KIT UPDATES CLASSIC DYNACO PREAMP

DIY and vintage audio upgrades received a boost with Updatemydynaco's regulated power supply upgrade kit for Dynaco's PAT-4 Preamplifier.

According to Dan Joffe, of Updatemydynaco, the regulated power supply circuit board replaces the old silver capacitor can while reusing the same mounting holes.

Updatemydynaco, a product line of



Updatemydynaco now makes a regulated power supply upgrade kit for the PAT-4 preamplifier.

Akitika, makes upgrade kits for Dynaco's classic solid-state audio equipment. A range of enhancements are available for the Stereo 120 power amp and PAT-4 preamplifier. For more information, visit www.updatemydynaco.com.

SINGLE-ENDED KT120 AMPLIFIER

True Audiophile, an exclusive US importer for Audion Tube Audio, introduced what it describes as the world's first single-ended KT120 amplifier at T.H.E. Show in Newport Beach, CA (May 31–June 2, 2013). The Audion Super Sterling 120 uses Tungsol KT120 pentode tubes. The amplifier delivers



The Audion Super Sterling 120 is a single-ended KT120 amplifier.

approximately 24 W into an 8-Ω load. The amplifier has been designed to work with lower-efficiency loudspeakers from approximately 86 dB and up. This product marks Audion's move to less efficient speaker dependency for its amplifiers. Audion designs and manufactures the transformers and chokes. Even the capacitors are made to Audion's exact specifications. To ensure quality, the chassis are finished in-house, including the powder coating and electroplating. Every component is hand built, usually point to point. The Super Sterling 120 will also be available in a parallel single-ended 40-W amplifier and a 60-W push-pull amplifier. Both are mono blocks. For more information, visit www.trueaudiophile.com

B&C ADDS NEW HIGH-FREQUENCY DRIVERS

B&C Speakers officially released several new and interesting products at the

ProLight + Sound show, held in Frankfurt, Germany (April 10–13, 2013).

The updated range of 75-mm (3") voice coil high-frequency drivers are particularly noteworthy. These drivers feature a robust titanium diaphragm that incorporates next-generation surround geometry with a new, optimized phase plug. Significant research has yielded a new coil former that solidifies the diaphragm with negligible increase in mass. The result is improved high-frequency linearity and reduced distortion. The 1.4" exit DE90TN (ferrite) and the DE980TN (neodymium) provide a solution for two-



B&C's new DE980TN high-frequency driver shown here is just one of the new models B&C officially launched at the 2013 ProLight + Sound show.

way point source enclosures, as well as for mounting a waveguide horn in multi-driver line array systems. The 2" exit DE985TN (neodymium) is also available.

The DE14 and the DE14TN are considered to be the next evolution of the industry standard DE12, a 1" exit ferrite magnet high-frequency driver. The 44-mm (1.7") diaphragm driver features an optimized phase plug and rear cap that improve frequency response with lower distortion. Finally, the new DE254TN, 44-mm (1.7") voice coil, titanium diaphragm high-frequency driver offers an excellent value in a 1.4" exit driver.

All new high-frequency driver models are now in production. For more information, visit www.bcspeakers.com

SAGE ELECTRONICS INTRODUCES A D.I. BOX

sage ELECTRONICS now offers a direct input (D.I.) box called the SE-D.I.3 Mighty G. Designed by Quentin Meek of QZIC Engineering, in partnership with sage ELECTRONICS's founder, Phillip



The MIGHTY G is a pocket-size D.I. box that plugs directly into most acoustic and electric guitars, basses, and keyboards.

Victor Bova, the MIGHTY G is a palm-size D.I. box featuring new old stock (N.O.S.) vintage germanium transistors.

The MIGHTY G is 3.5" long and 1.5" wide. It is housed in a showroom-finish die-cast enclosure. It fits in your pocket and plugs directly into most acoustic and electric guitars, basses, and keyboards. It has been designed to sound great, look important, and perform flawlessly in live and in-studio settings.

The unit is phantom powered. The input connector is a Switchcraft 0.25" male-switching jack. There is also a female input model (the SE-D.I.3 F MIGHTY G) for use with instruments with hard-to-get-to output connectors. The truly (not quasi) balanced output jack is an all-metal Switchcraft gold-plated three-pin XLR connector.

The MIGHTY G's germanium transistors contribute a sonic presence not heard in passive transformer and integrated circuit (IC) direct-box designs.

The MIGHTY G features Class-A active electronics (no ICs) and hand-tested and matched vintage (N.O.S.) germanium transistors. The direct insertion (i.e., plugs directly into your instrument), eliminates signal loss, which lowers hum and interference in both single-coil and humbucking pickups. It is phantom powered (no batteries required), with symmetrically balanced outputs (not quasi balanced). A silent plug built into a 0.25" jack stops hums, squeals, and pops when the plug is removed from jack.

The SE-D.I.3 MIGHTY G costs approximately \$300. It is available from the manufacturer. For more information, visit www.sageelectronics.com. aX

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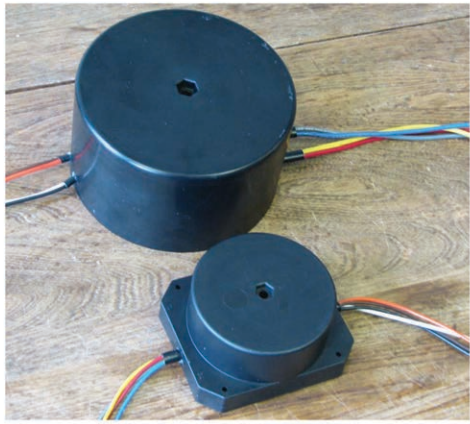
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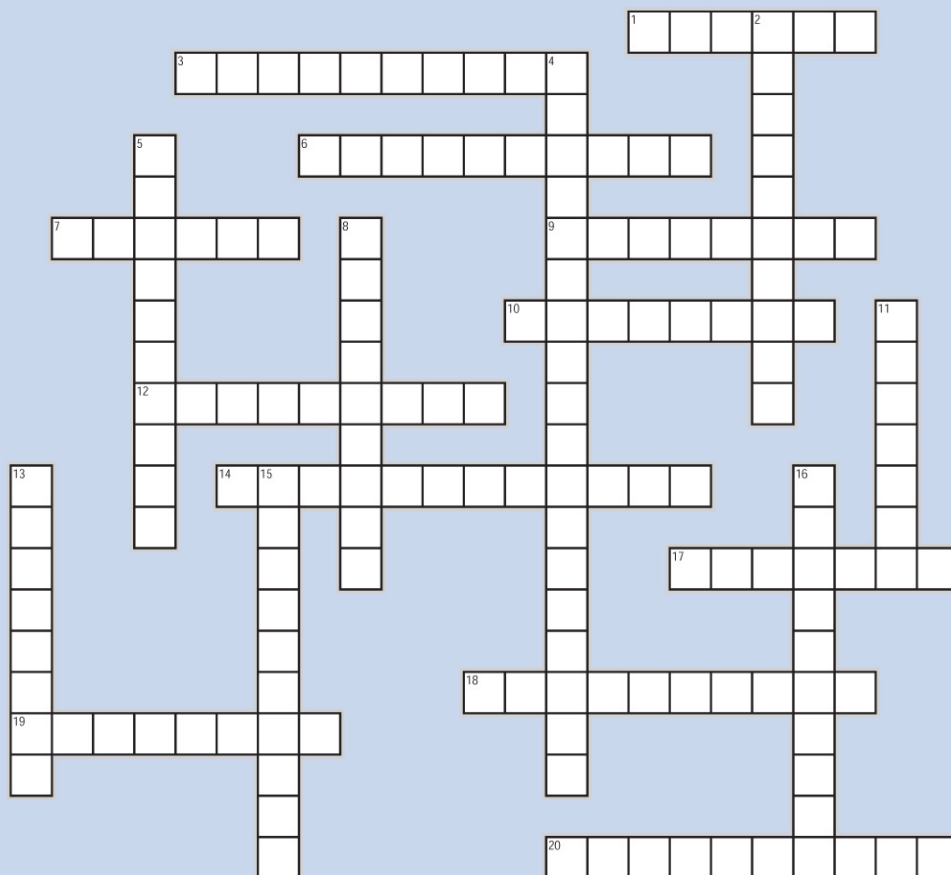
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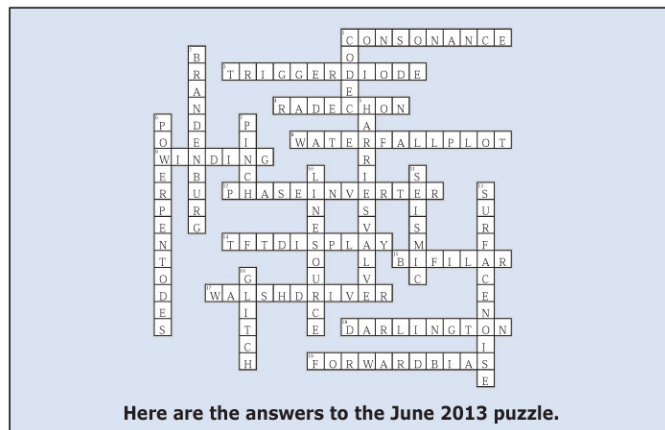
Editor's Note: Answers to this puzzle are posted at audioxpress.com/crossword and will be available in the next issue.

ACROSS

1. An early tube rectifier
3. Outputting its maximum [two words]
6. Recording technique that creates 2- and 3-D sound fields
7. Conducts electricity [two words]
9. Makes intense high-frequency energy
10. Absorbs sound or electromagnetic wave reflections
12. Early capacitor [two words]
14. Used to break the connection between the earth and the instrument chassis [two words]
17. Glass tube used in receivers
18. Detects magnetic anomalies
19. We use them to transmit vibration
20. Human ears' ability to localize sound [two words]

DOWN

2. Tweaking to get the best amplitude level [two words]
4. Modifies signals, making them louder or softer [three words]
5. Often denoted by L
8. English audio engineer (1921–1995) who revolutionized audio analog electronics
11. This type's elements include cathode, plate, control grid, screen grid, and suppressor grid
13. Helps generate or amplify microwaves
15. Occurs in ferromagnetic materials
16. Includes a G key [two words]



Here are the answers to the June 2013 puzzle.



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CONTRIBUTORS

Bill Christie ("Build an External Phonograph Preamplifier," p. 12) developed an interest in electronics at the age of 6 after his father helped him build his first crystal set. Since then he has built numerous kits. Bill started building amplifiers from scratch while he was still in high school, starting with a mono tube amp designed by Mullard. He graduated from the University of Akron in Ohio with a BS in Physics. Bill is a retired electrical engineer from Goodyear Tire & Rubber Company who spent 35 years designing controls and instrumentation for manufacturing.

Vance Dickason ("Beston's Horn-Loaded Transducer," p. 37) has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—which is now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is the editor of *Voice Coil: The Periodical for the Loudspeaker Industry*, a monthly publication. Although he has been involved with publishing throughout his career, including receiving an AES Citation Award for his ongoing contribution to audio education, he still works as an engineering consultant to a number of loudspeaker manufacturers. This currently includes Samsung, The AVC Group (Niles and Sunfire), Artison, and Emotiva.

Gary Galo ("Parasound's New Zdac," p. 28) retired in October 2010 after 34 years as an Audio Engineer at The Crane School of Music, SUNY, in Potsdam, NY. Not ready to quit entirely, he works half-time in that capacity. Gary is also a graduate of The Crane School of Music, where he earned a BA in Music Education in 1973 and an MA in Music History and Literature in 1974. He is the author of more than 260 articles and reviews on both musical and technical subjects. An active member of the Association for Recorded Sound Collections (ARSC), he was co-chairman of the ARSC Technical Committee from 1996–2004 and the Sound Recording Review Editor of the *ARSC Journal* from 1995–2012. He has been a regular presenter at ARSC conferences and several of his conference papers have been published in the *ARSC Journal*. Gary is a regular contributor to *audioXpress*.

Dr. Richard Honeycutt ("Output Transformers," p. 34) has been an audio enthusiast, musician, recording engineer, broadcast radio engineer (First Class Commercial licensee since 1969), radio announcer, electronics repairman, tube guitar amplifier designer, speaker system designer, pro-sound/video designer, and college electronics teacher. He is now a consultant in acoustics and electroacoustics. He earned a PhD in Electroacoustics from the Union Institute, and lives in Lexington, NC, with his wife, Betty Jane, near his daughter Alyson, her husband, and two of his three grandchildren.

Mike Klasco ("High-Performance Speakers Meet the HD-Voice Challenge," p. 8) is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike specializes in materials and fabrication techniques to enhance speaker performance.

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Steve Tatarunis ("High-Performance Speakers Meet the HD-Voice Challenge," p. 8) has been active in the loudspeaker industry since the late 1970s. His areas of interest include product development and test engineering. He is currently a Support Engineer at Listen, in Boston, MA, where he provides front-line technical support to the SoundCheck test system's global user base.

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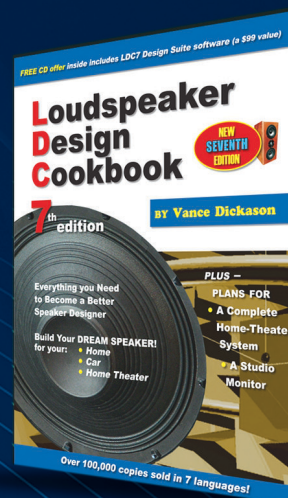
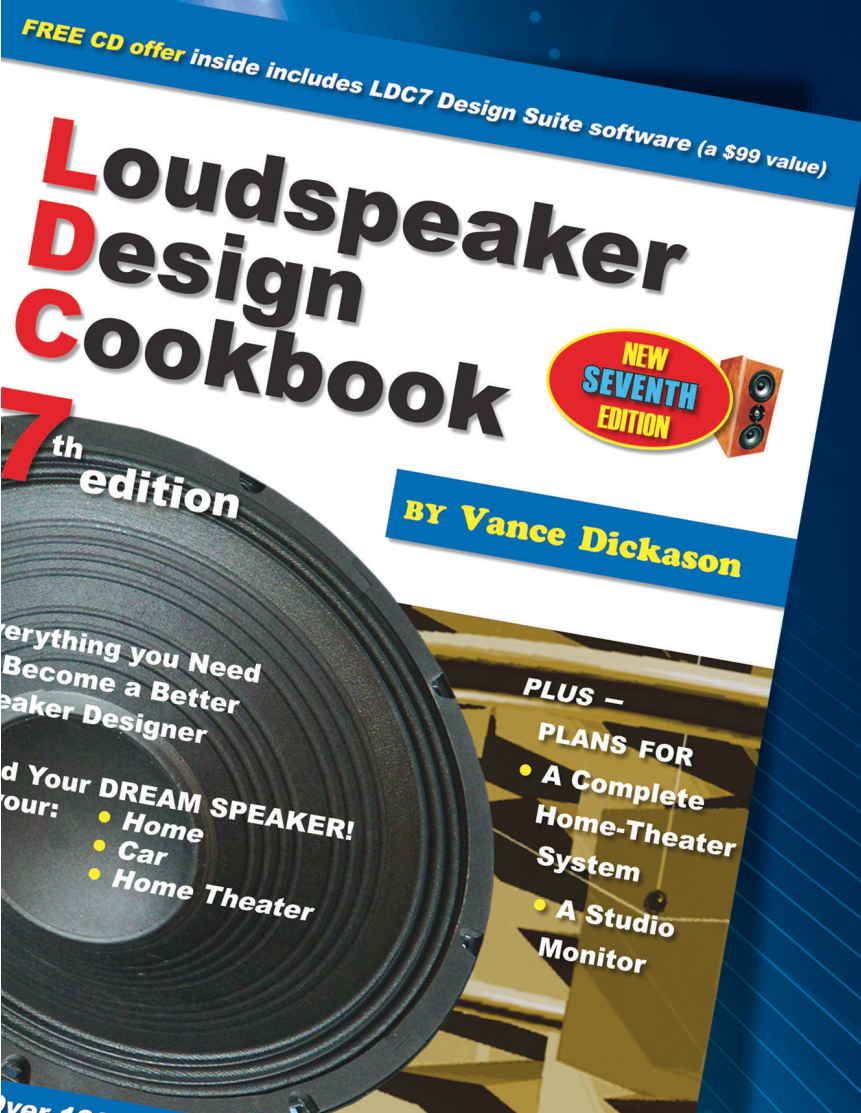
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